Recording
ENGINEER/PRODUCER

The Applications Magazine for Audio Professionals

$4.00

AN·INTERTEC PUBLICATION

Live Sound
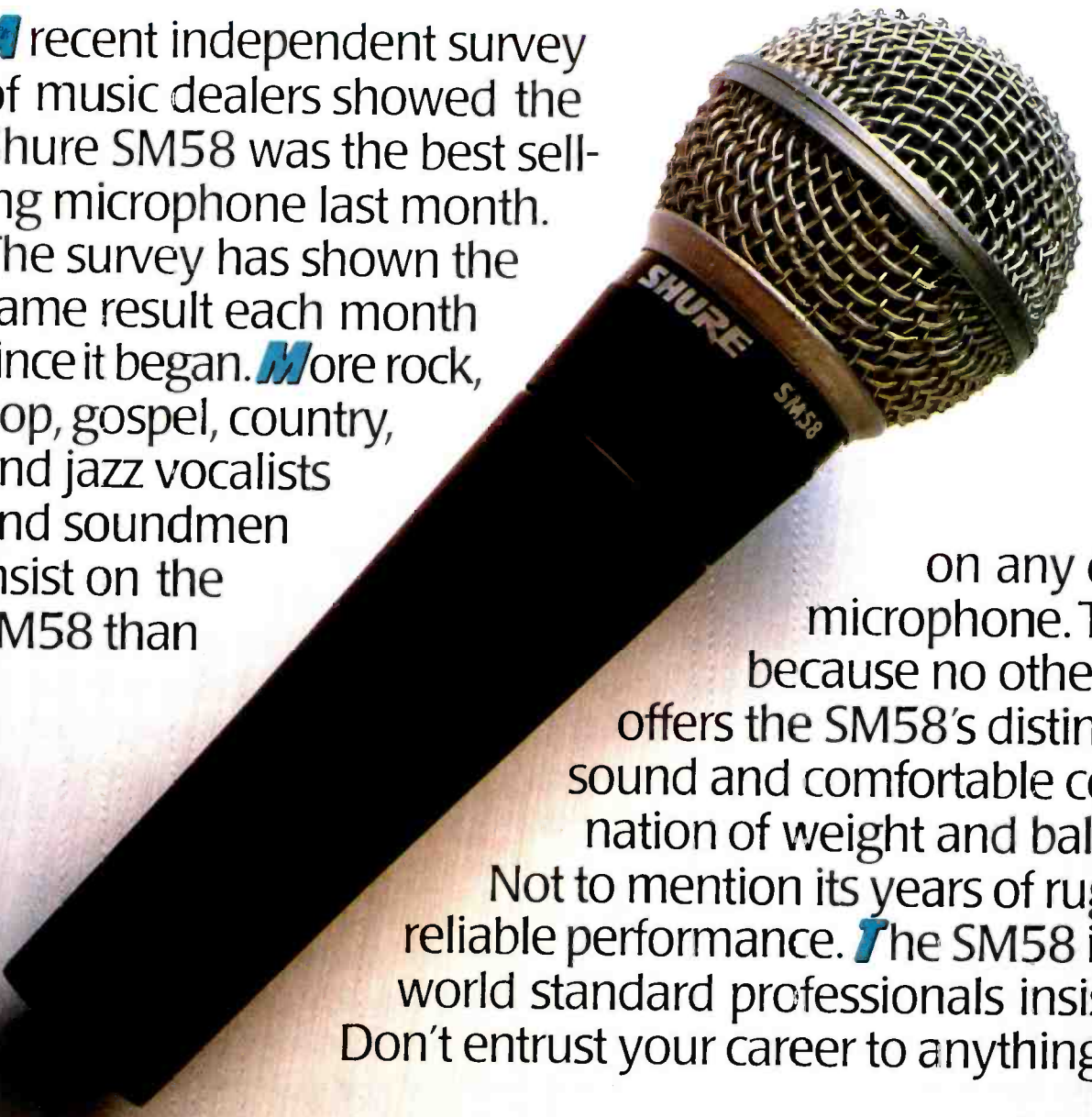# Reinforcement

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A concert at the St. Paul Civic Center exemplifies live sound reinforcement's special challenges—and rewarding results. Photo courtesy of JBL.

EDITORIAL

A "Sound" Education

While attending the Sixth International AES Conference on sound reinforcement in Nashville recently, I was impressed by a particular statement made by Clifford Henricksen of Electro-Voice. Essentially his point was that as a society, we take almost no time to educate ourselves about sound.

He went on to note that although we teach our children to use the alphabet, manipulate numbers, and recognize and use colors, along with a wide variety of other useful bits of information (all of which are means of visual communication), there are almost no general education courses on sound or acoustics.

For early man, it's safe to say, the sense of hearing was extremely acute and was essential to survival. Along with the sense of smell, it provided an effective early warning system, a means of communication and possibly a pathway for entertainment as it does today.

As the human race evolved, other methods of communication became possible or popular such as written languages. And though our biological capacity for aural reception has not deteriorated, the level of dependency, and therefore the level of acuteness and awareness, has been on a steady decline.

As we enter the 1990s, our regard for audio quality is once again growing. And as the quality and technology of audio products develop, a new awareness is becoming evident. Notice I use the word *evident*, because this is not a new phenomenon, but rather a reawakening of the significance of one of mankind's primary senses.

As a professional, have you ever noticed how average people react when asked about sound or sound systems? Generally, they like just about anything with a reasonable amount of power (intelligible or not). But beyond that, they have little or no vocabulary for explaining what they like or don't like about the sounds they hear, or an informed basis for understanding what is influencing the differences they do hear.

Why is this necessary?

Perhaps the main reason poor sound quality is accepted by the vast majority of people is they don't understand anything about the basic differences between good and bad sound. They don't have a clue as to what causes a system to sound good or bad, and if subjected to really bad sound, they haven't learned that there are solutions to that problem as well.

Is this really possible? Let me cite a few examples. The paging systems in bus, train or airport terminals. The public address systems in many stadiums and auditoriums. Or, the sound quality most of us have grown up with while watching television. Some systems such as these are so bad that no one can understand what is being broadcast—even program material as basic and familiar as the human voice. Yet as a society, we often accept these highly distorted means of communication as fact of life.

Who's to say what is good or bad? That's a good question, but the point is wrong. What Mr. Henricksen and I are suggesting is education to raise the level of awareness and the ability to communicate in basic audio terms. I think it's safe to say that differentiating between good and bad (be it manners or sound) can be learned. By elevating awareness through education, we prepare the populace to make their own informed evaluation of sounds to which they are exposed.

What can be done?

Perhaps some basic education on the elementary, secondary and collegiate level that teaches students to recognize some of the various characteristics of sound, the basic tonal range of a variety of instruments and voices, and a fundamental understanding of acoustics. This would go a long way in helping the average citizen better understand the nature of sound and the impact of the acoustic environment in which it is presented.

As professionals, we can insist the sound quality we provide is the best possible. This means putting an extra effort into educating our clients about the value of clear, crisp sound—and the effect that various types of acoustic environments have on sound. We can speak out and register formal complaints whenever we are exposed to drastically inferior sound reproduction. As members of various audio societies, we can institute and support efforts to represent this and other audio-related themes at the state and federal level. And, we can teach our own children the characteristics and value of good sound—so they may grow up fully appreciating the efforts of audio professionals, and the delights of the auditory sense.

Michael Fay
Editor

RE/P

LETTERS

Digital issues

From: Brad S. Miller, chairman and CEO, By The Numbers, Incline Village, NV.

In the March issue, which features digital audio technology, there are three issues raised that require further scrutiny.

Jerry Whitaker writes on page 60 ["Disk Recording Technology"]: "Although digital recording is, theoretically, error-free, much discussion has been given to error correction for various digital recording formats. In point of fact, there is no such thing as error correction. More correctly, it is error concealment."

The next time that you attempt to use your ATM card when you believe that you have X dollars on deposit, you had better not rely on the concealment of errors to withdraw your money!

The second and third issues are contained on page 65 from Tom Martin ["The Capital Reinvestment Zone"]: "The higher the sampling rate, the finer the resolution, and, therefore, the better the sound. Further, no error correction is needed in random access recording as there is with recording tape."

Unfortunately, the foregoing is a commonly held misconception. Resolution is determined by the number of slices (16 bits = 65,536, 18 bits = 262,144 slices), and not by the sample rate. The sample rate determines *only* the highest audio frequency to be sampled (Nyquist rate 2 × Fs, when Fs = the highest audio frequency to be sampled).

Error correction, in general, is implemented in most systems to maintain the short-term validity of the recorded data. It is, in fact, the reconstruction of the original data. That is why the ATM must not conceal errors, or you would be hard-pressed to recover the correct (original) amount of your deposits.

Of course, there are differences in the robustness of error correction schemes, depending on their design. Meanwhile, the controversy rages on, and we shall see if, when and where the other shoe drops.

Tom Martin replies:

The word "resolution," in the strictest sense of terms, technically can refer to the "number of slices" (16 bits = 65,536, or 18 bits = 262,144). However, I think the real meat of my statement was *the higher the sample rate, the better the sound*. My concern was and is with the sound quality, which is closely related to the sample rate.

A human voice sampled at 6kHz does not have the same audio "resolution" as one sampled at 50kHz (in essence, voltage levels are checked 6,000 times per second vs. 50,000 times per second). In our listening test (performed with a Synclavier) there was a clear difference between certain materials sampled at 44.1kHz vs. 50kHz, the difference showing up as a lack of harmonic content that was available in the upper frequencies. This contributed to an overall difference in the timbre of the sampled sounds; an idling jet on the airport tarmac did not have the same "transparent" sound when sampled at 44.1kHz.

Sydney Alonso (director of research and development for New England Digital) states that although the ultra-high harmonics are inaudible by themselves, their effect is quite profound on our perception of the audible audio spectrum. This is the area of psychoacoustics. The Synclavier's anti-aliasing stopband filters cut off at 44.1kHz. I guess what all of this still means is that the higher the sample rate, the better the sound.

Obviously, with regard to better sound, other factors enter into the scenario, such as a system's digital/analog converters, slow slewing transistors or inferior LC filter networks. These all must be taken into account before we address 16-bit vs. 18-bit vs. 24-bit or sample rate.

As to your second objection to my statement that "no error correction is needed in random access recording," I'm sure you will agree with me that one is bound to encounter more errors when the record/reproduce method is a piece of magnetic recording tape (a storage medium with surface imperfections) being dragged across some metal vs. a rotating magnetic disk with the recording/reproducing element ideally never touching the storage medium (the disk). Common sense would seem to indicate that more error correction would be necessary with the tape medium. The error correction for the Synclavier is hundreds of times less than that needed for digital tape recorders.

My article was aimed at the guy who has to pull his wallet out and pay for some technology. One of my major complaints with the industry is that the manufacturers of this "leading edge" technology, in an effort to distinguish their products from others, will use what I call the "half of a half" or "po-ta-to/po-tah-to" logic. However you cut it, it's still a potato. It's

unfortunate that technical data as it relates to modern recording equipment has become so overwhelming that the average studio owner must hire an engineering consultant just to pour over a manufacturer's product literature and translate it into layman's language, as well as relate and compare it to another competitive product.

Microphone terminology

From: John J. Thomson, production sound mixer, Flying Tiger Productions, Toronto.

Re: Bruce Bartlett's letter in the April issue re: Fred Ginsburg's article on production sound [November 1987]: In Mr. Bartlett's attempt to clarify certain points, I believe he has further muddied the waters, which stems, in part, from Mr. Ginsburg's original use of the language.

His interpretation of Mr. Ginsburg's terms for types of lavalier mics, "transparent" and "proximity-oriented," as meaning omnidirectional and unidirectional is, I believe, incorrect. As quoted in the letter, "Until recently, all professional-grade electret lavaliers were proximity-oriented—in other words, they tended to add presence to close dialogue while rejecting background ambience."

Mr. Bartlett takes this as evidence that the mics must be unidirectional. However, an alternative interpretation may be that Mr. Ginsburg may be talking about sub-categories of omnidirectional lavs. The rejection comes from the "presence" boost, which tends to bring the voice forward and out of the "background ambience." It used to be true, as Mr. Ginsburg states, that all omni lavs displayed this, but recently some new omni lavs have come on the market, such as the Sennheiser MKE-2 and Sonotrim, with noticeably less "presence" boost. Thus, they can be called more "transparent."

Indeed, this must be the case, as unidirectional lavs are generally not used in production sound, because the very ports that make them unidirectional, of necessity, tend to become covered when the mic is hidden in the costume of the actor. One is no further ahead and might as well use an omni, because they tend to sound better than the unidirectional lav.

With regard to "electret-condenser shotgun mics," I do not doubt the truth of Mr. Bartlett's statement about sensitivity. However, I think the point is that no professional sound recordist would use one,

NEWS

Universal adds satellite recording

Universal Recording, Chicago, has added satellite recording capabilities, which it calls Stereo Digital Satellite. The service is available between Chicago, New York and Los Angeles, and is targeted to advertising clients for playback and overdubs. Specific talent in any of the three cities can be recorded elsewhere, which eliminates the need to fly talent in to the city.

Nakamichi develops R-DAT transport

Nakamichi has developed an R-DAT transport that it says provides more precise tape guidance and higher search speeds. Called FAST, for fast access stationary tape guide transport, the mechanism features a half-load position that provides program search at 400 times playback, twice as fast as other DAT transports.

Other advantages, according to the company, include faster startup, two seconds after the command is given, and the use of stationary tape guides at both sides of the head drum.

Compatible with the DAT Conference standard, the transport will be used in Nakamichi's upcoming line of DAT decks. The company has also entered into an agreement to supply the transport to other companies.

Otari sets prices, moves headquarters

Otari has announced a price increase on all of its products and has made major changes to its distribution pricing policies. Effective May 1, the price increases were to compensate for the weakening dollar. Otari also will no longer publish a list of prices that float above what people actually pay. Instead, it will publish a professional user price, which the company expects will be much closer to the street price.

The company also was scheduled to move into its new headquarters in Foster City, CA, in early May. The building is double the size of its previous offices and will include a studio listening room in addition to offices, warehouse, quality control and service areas.

Harman International acquires Soundcraft

Harman International Industries has signed an agreement for the acquisition of Soundcraft Electronics, Ltd. The acquisition is expected to be completed in June. Soundcraft manufactures automated multitrack, compact, concert, and sound and broadcast consoles. JBL Professional, a division of Harman, has been the exclusive U.S. Soundcraft distributor since January 1986.

News notes

New England Digital has signed a remarketing agreement with Columbine Systems, a developer and marketer of traffic, sales and accounting software for radio and TV stations. Columbine will distribute the Synclavier and Direct-to-Disk system to its domestic and Canadian customers. The move is part of NED's entry into the broadcast radio market.

GC International has acquired Capitol Industries-EMI's lacquer audio disc master manufacturing plant in Winchester, VA.

Ampex has introduced a new packaging system for its 600 series industrial audiotape, which underscores recent performance improvements and establishes a uniform packaging look.

Imax Systems Corporation has acquired a 51% interest in Sonics Associates.

Compact disc orders to Shape Optimedia surpassed first quarter projections, and the company expects its 1988 market share to increase by more than 75%.

Integrity Audio, Atlanta, a new company for the sales and service of Amek/TAC consoles in the Southeast, is a partnership between Studio Supply Company, Nash-

NEWS

ville, and Lewis Frisch, previously the manager of Showcase Audio in Atlanta. Integrity's address is 2759 Skyland Drive NE, Atlanta, GA 30319; 404-636-2601.

Mitsubishi Corporation of Tokyo has acquired a substantial interest in Electro-Sound Group. The two companies are joint shareholders in Memory-Tech, a CD producer based in Plano, TX.

Marshank Sales has moved to 6733 S. Sepulveda Blvd., No. 190, Los Angeles, CA 90045; 213-670-7071.

WaveFrame Corporation has appointed six new distributors in Europe: Syco Systems, United Kingdom; Spye Srl, Italy; Music-Land, France; Amptown Electoacustic, West Germany, Synton, Benelux; and New Musik, Denmark.

Klark-Teknik has purchased Celco Inc., the U.S. distributor for Celco Ltd., a U.K.-based lighting manufacturer.

The Joiner-Rose Group has opened a New York office, which will provide audiovisual and television system design as well as architectural support. The office is located at 8-10 W. 19th St., New York, NY 10011; 212-633-1759.

John Fluke Manufacturing has extended the warranty period on its models 25 and 27 multimeters to three years.

Akai clinician Roger Linn recently presented an in-depth seminar at West L.A. Music on the Akai MPC60 MIDI Production Center. The unit is a result of a joint venture between Linn and Akai. About 200 people attended the seminar.

The first seminar in Japan covering Quested Monitoring Systems was recently presented in Tokyo by **Soundcraft Japan Ltd.,** the Japanese importer and distributor. Roger Quested covered his background and the design philosophy behind the monitors.

DOD Electronics has presented a variety of sales awards. MBT Associates, Pittsburgh, was awarded for most improved territory sales. Crescendo and Associates, Miramar, FL, was awarded the DOD Rack Signal Processor Sales award and the Audio Logic Sales award. Elliot Goldman, the company's New England sales representative, was awarded the FX Product Sales award. Gary Castelluccio Associates, Clifton, NJ, was awarded the DigiTech Product Sales award, the Sales Representative of the Year award and the Gold Club award.

Digital Audio Research has sold the first SoundStation II to Finesplice Ltd., an editing and post-production facility located in Harmondsworth, West Drayton, England.

LaserDisc Corporation of America is relocating to Long Beach, CA, effective Sept. 30. The company is a subsidiary of Pioneer Electronics and will set up offices in Pioneer's U.S. headquarters.

CDT Audio has been purchased by Bruce A. Forbes of Forward Technologies and Bob Reams of Paranamics, and has been renamed Paranamics Audio Division. Forward Technologies will continue to handle marketing and distribution in the United States.

Ramsa has named New West Audio, Burbank, CA, as the company's representative of the year for 1987.

Total Audio Concepts is now offering a 2-year parts and labor limited warranty on all new products sold in the United States, including the Scorpion, Matchless and SR-9000 consoles.

Meyer Sound presented a commemorative UPA-1A loudspeaker to Autograph Sales, London, at the Paris AES in March. The loudspeaker represented the 1,000th UPA-1A that Autograph has purchased.

Amber Electro Design has moved to 3391 Griffith St., St. Laurent, Quebec, Canada H4T 1W5; 514-735-4105 (800-361-3697 in the United States).

The Mitek Group has begun construction on an additional 61,000 square feet of manufacturing space adjacent to its current facility, which will increase its total facility to 143,000 square feet.

Sound Ideas has appointed Studer International as the worldwide distributors for the company's music production and sampler libraries. Sound Ideas will continue to distribute directly in North America and Australia.

People

Howard Schwartz, president of Howard Schwartz Recording, New York, has been named to the board of directors of two industry associations, the Society of Professional Audio Recording Services (SPARS) and the International Teleproduction Society/New York (ITS/NY).

Bose has announced two appointments in its professional products division. **Tim Dorwart** has been promoted to national field sales manager. **Robin M. Sibucao** has been named eastern regional sales manager.

Adrian Weidmann, international manager of Bruel & Kjaer's pro audio group, has relocated to the company's U.S. headquarters in Marlborough, MA, to concentrate on the U.S. market.

Peter Flicker has been named vice chairman of the Neve board of directors.

Otari has announced two appointments. **David Ruttenberg** has been promoted to central region sales manager. **Pierson Ball** has joined the staff and will work with dealers, representatives, sound contractors and end-users in Central and South America.

Klotz & Company has appointed **Brian Latham** as sales director in the United Kingdom.

Brad Friedrich has been named director of marketing for the magnetic products division of Fuji Photo Film U.S.A.

Mike Halleck has been named southwest regional manager for Studer Revox America.

Marty Blanchard has been promoted to senior market research and planning analyst at Ampex's magnetic tape division.

Don C. Klick has been named vice president of operations for Centro.

Brian L. Bristo has been named to the Chicago office of AudioLine.

Mark R. Nixon has been named vice president of sales and marketing for Ultimate Support Systems.

Correction
In the March issue, the phone number to order the "Master Tape Preparation Guide" from Diskmasters was incorrect. The correct phone number is 1-800-468-9353.

MANAGING MIDI

By Paul D. Lehrman

Unclogging the Sync

Folks want to sync to tape for a variety of reasons: to use sequenced music along with vocal or acoustic-instrument tracks, to build thicker textures from a small number of synthesizers, and to lock music to pictures on videotape or film. But linking MIDI sequencers with tape-based systems is one of the most confusing aspects of MIDI.

The problem is as old as MIDI itself (five years this month), and most of the early methods have had serious drawbacks. But things are getting better, and the ultimate solution may finally be at hand. The original MIDI spec included two provisions for syncing sequencers and drum machines with the outside world and each other: MIDI Timing Clocks and Song Position Pointer (SPP). A Clock is a single-byte message that occurs every 1/24th of a quarter note, and serves as a real-time reference; when a sequencer receives a series of Clocks, it knows the desired tempo.

The SPP tells the sequencer *where* to start playing. SPP is a 3-byte message, with one command "*status*" byte and two data bytes. The SPP number represents the number of Clocks from the beginning of the sequence, divided by 6, which is the same as the number of 16th-notes since the beginning. The two data bytes are seven bits each, which gives 16,384 possible values. Because a 4/4 measure contains 16 16th-notes, an SPP message can locate anywhere in a sequence up to 1,024 measures long—slightly more than 8 minutes

Paul Lehrman is *RE/P*'s electronic music consulting editor and is a Boston-based producer, electronic musician and free-lance writer.

at a metronome setting of 120bpm. When you are using two MIDI devices, syncing is a snap. With tape, life is a little more complex. Unfortunately, MIDI signals cannot be recorded as audio, so there is no way to get Clocks and Pointers on tape. One way around this is to translate the Clocks into a frequency-shifted audio signal: Send Clocks into a hardware box generating a sine wave at, say, 2,000Hz, and every time a Clock is read, the frequency shifts by 250Hz for a few milliseconds. This signal will record just fine, and when played back, the same box can read the frequency shifts and generate Clocks the sequencer can lock to.

There are two problems with this. First, there's no location information stored on the tape; if you start the tape in the middle of the sequence, the sequencer will start from the beginning. A couple of companies have developed a solution for this: "smart" FSK, in which location information is encoded into the FSK signal itself. When the hardware box reads the FSK signal, it checks the location first, and sends out a Pointer message before it starts sending Clocks. The second problem is that MIDI Clocks are related to tempo; they are generated at 24 Clocks per quarter-note, not per second. If you always use the same tempo in every sequence this is no big deal—but most musicians don't. Striping a tape with FSK involves actually playing through the sequence once while the MIDI-to-FSK converter creates the sync signal. Once the signal is on tape, it cannot be altered, which means no tempo or length changes without restriping.

For many users, this is no big deal; there are generally no tempo changes in a basic 30-second jingle. But, for more complex compositions, this can be a drawback. And there are plenty of users who need a more standard form of sync, specifically SMPTE time code. Enter the SMPTE-to-MIDI converter. This device reads SMPTE code. When it first receives a time code number from the tape, it converts it into an SPP, and as the tape runs, it generates MIDI Clocks. SMPTE time code contains no tempo information, only real-time position. The converter has to generate an SPP whenever it receives a SMPTE number, and when the tape is running, it has to generate Clocks at the right tempo. To do that, it has to know the entire history of the sequence, in terms of the tempo value

at every beat, up to that point in time. This history, along with the SMPTE start time (or "offset") for the beginning of the sequence, is contained in the memory of the converter as a "tempo map." A typical tempo map might specify a start time of 01:45:33:12, a tempo of 120bpm for 25 4/4 measures, 160bpm in 3/4 for measure 26, 92bpm in 6/8 for measure 27, etc.

Unfortunately, the user interface for designing the tempo maps on a hardware converter often leaves much to be desired—in fact, few of them are any better than a drum machine. If there are a lot of tempo changes in a sequence, programming them one by one with a limited set of multifunction buttons can be a major pain. In addition, many of the converters have limited memory capacity and no easy way to off-load the memory.

One way partially to overcome this is to allow the converter to "learn" a tempo map from the sequencer in real time; you design the map in the sequencer and then play the sequence back into the converter in real time.

But the best solution, which is just now being implemented, is to allow tempo editing within the sequencer, with the edits taking effect immediately without the need for "restriping". This can be done by converting SMPTE to some other form of MIDI sync that is *not* tempo related. A couple of manufacturers have tried this using special timed system-exclusive messages, but there is now a better solution: MIDI Time Code.

A sequencer that can recognize MIDI Time Code, which is not much harder to do than recognizing Clocks and Pointers, will convert it *internally* to location and tempo information, and so the converter needs no controls at all, and there's no tempo map to worry about. There is no need for any communication between the sequencer and the converter, except the MIDI Time Code itself. Getting a sequencing program to deal with MIDI Time Code is not a terribly difficult coding job, and most major software sequencer developers are already doing so. As long as there are different ways to do it, there will be confusion regarding tape-to-MIDI sync. But, the adoption of MIDI Time Code should do for the MIDI studio what the standardization of SMPTE time code did for the tape-based studio: make life a lot easier. RE/P

TOTAL AUDIO BRILLIANCE

SPARS ON-LINE

By John Eargle

Making It to the Top

While most aspiring recording engineers dream of fame and success in the studio, it is a simple fact of life that only a few will make it to the top. Skill is a prerequisite, but so are patience, determination and just plain luck. Enterprising engineers don't wait for that stroke of luck; they make opportunities for themselves. They may become mastering engineers, or specialize in some other phase of post-production to develop skills that are complementary to those of the studio owner, but no less essential to the industry.

Many engineers develop these skills as career backups while waiting for that opportunity, in a star/understudy context, to show what they can do at the console.

It is a good bet that most engineers have never considered the role of the live sound mixer as an alternative to the studio, or as a parallel path of development. It will be instructive to look at this industry in detail to see the inevitable comparisons with studio work.

In contemporary pop/rock recording, the studio engineer gets involved at the beginning of the creative process. The roles of mixer, producer and artist are not mutually exclusive, but it is basically the mixer, at least in sonic terms, who shapes the product as it comes into being over the control room monitors.

The music reinforcement mixer comes on the scene much later, usually after the group has had success with its recordings. In fact, many tours are partly underwritten by record companies, in an effort to stimulate record sales. In these cases, the public expectation is that the group on

John Eargle is president of JME Consulting, an author and recording engineer based in Los Angeles.

tour sound substantially like its recordings. So, what the live sound mixer is doing is closely related to a mixdown session in the studio. The house mix is given the benefit of all the signal processing that may have gone into the recordings.

Both the studio and reinforcement engineer must have a good feel for their respective acoustical environments. In the studio, the mixer is concerned about microphone selection and placement, interference problems, proximity effect and early reflections. The live sound engineer is more apt to be concerned with the acoustics of large spaces (indoor and out), effects of delayed sound and of wind and temperature gradients in outdoor settings.

The live music mixer has to get everything right the first time. There is no stopping and starting over.

Both disciplines rely on a substructure of technical and logistic support, with clearly defined responsibility if everything is to run smoothly. An experienced tour company can rig a very large venue and be ready for a sound check in about four hours. Things may be less hectic in the studio, but the need for smooth running, days at a time, is essential to the creative process. Obviously, maintenance is important in both areas.

The road equivalent to studio headphone monitoring is the stage monitor, or foldback system. In a performance environment, the musicians must hear themselves loud and clear. While headphones operate in the milliwatt range, the foldback system alone can be driven by several thousand watts. Today, it is not unusual to see systems with power complements in excess of 50,000W.

Both engineers must be intimately involved with music, and both must be willing to put in long hours with a commitment to quality. Most SPARS members are keenly aware of the importance of technical quality in today's competitive world of recording. Things are no different in live music reinforcement, and the systems in use today reflect the latest developments in high-level, high-quality equipment design.

It would probably surprise many studio engineers to learn the impact high technology has made on the music reinforcement industry in recent years. The field has evolved in the hands of specialists, many of whom have worked closely with manufacturers in the development of ad-

vanced components.

Just as Porsche and Mercedes Benz fold the knowledge gained in racing activities back into their passenger car development, manufacturers in the professional sound industry use concert sound reinforcement as a proving ground for adhesives and high-temperature/low-distortion transducer designs, many of which ultimately find their way into studio monitors.

The high-tech revolution also extends to electronics. SPARS members are well aware of the impact of digital signal processing and recording in the studio. In fact, much of the current health of the recording industry derives from that technology. In music reinforcement, digital continues to play an important role. Many "smart" control systems are digitally operated, and the combination of digital control and array theory offers new flexibility in the shaping of frequency response and loudspeaker directional characteristics.

Talented people often move from one creative area to another. I would like to point out two gentlemen who have established enviable reputations in the studio before taking up live sound reinforcement.

One of the earliest was Larry Levine, an engineer at A&M Records in Los Angeles about 20 years ago. He moved into the live sound area when Herb Alpert went on the road with the Tijuana Brass.

In more recent times, James Locke, a classical recording engineer with London Records, has gone on tour with Luciano Pavarotti, supervising the high-quality sound for which Pavarotti's concerts are noted. In both cases, the move has represented a broadening of engineering and creative activity rather than a fundamental shift in direction.

The true merging of both disciplines takes place at a live recording. Last year at Carnegie Hall, Jack Renner of Telarc Records captured the spontaneity of Liza Minelli's performances before three weeks of sellout crowds.

This is noteworthy because it successfully combined Renner's determination to use high-technology equipment on the recording side (and mixing the program direct to stereo) with the demands for proper house coverage.

So, give it a thought. At the very least, find out more about the art and science of live sound reinforcement. You may be surprised how creative it is and how closely related it is to what goes on in the studio.

RE/P

At the top of the charts

Ampex hits the top of the charts with Ampex 467 digital mastering tape.

We not only pioneered digital audio tape, we refined it. The result is Ampex 467, a tape that sets the highest standards for all digital audio applications. And it's available in all open reel and cassette formats, including the 80-minute cassette length.

More top performers record their hits on Ampex tape than any other tape in the world. While opinion may vary on what it takes to make a hit, there's no argument on what it takes to master one.

AMPEX

Ampex Corporation, Magnetic Tape Division,
101 Broadway, Redwood City, CA 94063, 415/367-3809

UNDERSTANDING COMPUTERS

By Jeff Burger

Storage Media

As promised, this month we'll take a look at the different types of storage media. Interestingly enough, the development of media used by the computer industry closely parallels that of the recording industry.

Back in the dark ages (a few decades ago), we commonly thought of computers as imposing monoliths with massive reels of spinning tape, like some twisted multitrack in a parallel universe. Indeed, this method of storage is still in use today, but largely only for specialized tasks. The drawbacks are the same as reel-to-reel tape—the media is linear rather than random-access and comparatively more difficult to handle.

At the beginning of the personal computer era, storage was painfully performed using a creation of the audio industry—cassette decks. I say "painfully" because, like its open-reel ancestor, access is again linear. Finding the right position was a lot worse than finding the beginning of your favorite song on cassette. Compounding this problem was the fact that cassette media is not highly reliable. During the last decade, the manufacturers of electronic musical instruments have wrestled with the same problem; the digital data that makes up many a great synthesizer patch or drum machine pattern has been stored on audio cassette in lieu of a better media at an affordable price.

Jeff Burger is RE/P's computer consulting editor, and is president of Creative Technologies in Los Angeles.

In its search for a more reliable, random-access storage device, computer designers created the floppy disk in the late 1970s. This format gets its name from the flexible magnetic disk that is encased in a protective jacket. The floppy disk is to a disk drive what a record is to a record player—media vs. playback device.

The disk drive has its own version of a spindle that automatically inserts itself into the center hole of an inserted disk. When the computer needs to access the disk, the disk is forced to spin. An opening in the

> ## Designers created the floppy disk in searching for a reliable, random-access storage device.

jacket of the floppy disk allows the electromagnetic head(s) of the drive to come within close proximity of the surface—to read and write data.

Floppy disk media have improved drastically in the past few years. Veterans will remember the 8-inch floppies in vinyl jackets such as the original Fairlight system used. Around 1979, the smaller 5¼-inch disks such as those used by the IBM PC, Apple II and Commodore 64 became popular. The first drives had a single head, which meant that only one side of the disk could be used without turning it over. Each side held approximately 150k of data. More recent drives include two heads, one for each side of the disk, which effectively doubles the amount of storage the operating system can access.

Floppy disks are most commonly categorized in two ways: single- or double-sided disks and single- or double-density disks. In actuality, all disks are manufactured with magnetic media on both sides. The difference is that double-sided disks are tested on both sides by the manufacturer. Nine times out of 10, the other side of a single-sided disk is perfectly usable; the manufacturer simply hasn't certified it. Because single-sided disks often cost less, it's reasonable to assume you can use them as double-sided. However, do this at your own risk.

The term "density" refers to how tightly the magnetic media is packed onto the surface of the disk. Higher density results in a greater number of formatted tracks,

and more data-storage disk capacity. Single-density 5¼-inch disks provide the 150k of storage required for the Apple II and Commodore 64 operating systems. Double-density 5¼-inch disks are used for the standard IBM PC 360k drives.

Some PCs can be fitted with 5¼-inch or 3½-inch double-sided 720k drives. The IBM PC/AT can store 1.2Mb on a 5¼-inch quad density floppy, which is twice the density of double-sided. Typically, 3½-inch double-density disks hold about 800k on the Macintosh, Amiga and Atari ST.

Recent innovations have produced a new floppy disk standard in the 3½-inch disk (sometimes called diskette) with a hard plastic jacket. Currently, 3½-inch disks are used in the Macintosh, Amiga, Atari and IBM PS-2, among others. These disks are not only smaller, but are more durable and typically hold 400k per side. A protective metal slide protects the opening that the read/write head uses to access the magnetic surface.

All floppy disks can be physically write-protected to prevent accidental erasure. The 3½-inch disks have a simple "switch" that determines this status. The larger format disks have a notch that the drive searches for when writing to the disk. If the notch is covered by an adhesive write-protect tab, the information on the disk cannot be altered.

Regardless of their size, several rules apply to the care and handling of floppy disks. They are very similar to the guidelines for handling recording tape. Don't expose them to magnetic fields like speakers, permanent magnets, tape machine heads

> ## All floppy disks can be physically write-protected to prevent accidental erasure.

or de-Gaussers. Don't touch the magnetic surface of the media. Dust is an enemy, as are heat, humidity, spills and bending.

While 800k of storage was big time a few years ago, today's applications can eat that up in no time. Enter the hard disk, with storage capacities ranging from 10MBytes to more than 100MBytes.

The basic concept is the same as that employed with the floppy disk. There is a spinning disk(s) surface divided into tracks and sectors and a read/write head

flying above it. Because the disk(s) itself is rigid and affixed to a spindle, the positioning of the read/write head has much greater accuracy than with floppy media. As a result, many more tracks are possible on a hard disk of similar size, which translates to more storage. Hard disks are also significantly faster because unlike floppies they are always spinning, thus reducing access time. [See "Hard Disk Recording Technology" in the March issue for related information.]

Ah, but every solution brings a new problem doesn't it? The temptation is to put just about all the programs and data you have on a hard disk. What happens if it crashes (computerese for crossing its arms and saying there's nobody home)? Many hours and dollars can go down the drain if the hard drive has not been

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"Density" refers to how tightly the magnetic media is packed onto the surface of the disk.

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backed up or copied on a regular basis.

Two methods are popular—tape streaming and multiple floppies. Tape streamers use special tape cassettes to make a linear copy of all the data on a hard drive. Should the hard drive crash, the data on the tape can be used to reconstruct the hard drive.

Because tape streamers are almost as expensive as hard disks, multiple floppies can be used to hold or backup different segments of data from a hard drive. But, this method can be quite time-consuming.

There is a way to minimize this hassle. Presumably, you have the originals of all your applications on floppy disk. Set aside a folder or sub-directory on your hard drive for storing all of your irreplaceable work and documents. Then simply back up that folder alone.

Certainly, there are several less-common storage devices. There are removable hard drives, Bernoulli boxes (large sophisticated floppy cartridges) and CD drives. We'll take a closer look at these down the road, as they gain popularity. Next month we'll examine DOS or Disk Operating System—the interface between you and your disks.

RE/P

Flying Arrays for Performances in the Round

By David Scheirman

A detailed look at the mechanics of hanging temporary speaker arrays.
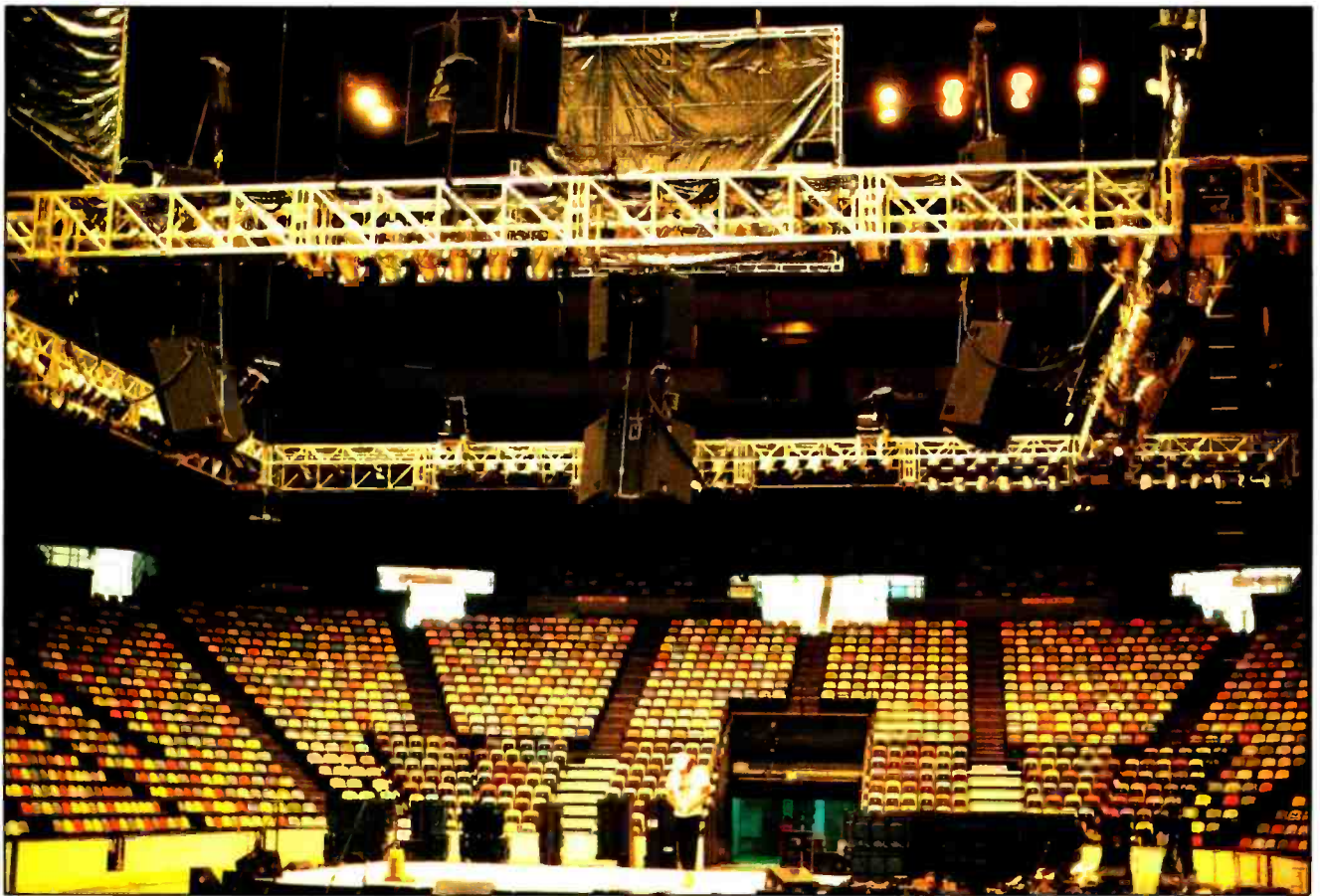
One of 1988's most financially successful tours was the nationwide road trip undertaken by Frank Sinatra, Dean Martin and Sammy Davis Jr. These three entertainers, with more than a century of stage experience, sought to expand their individual intimate cabaret-style shows into one large arena-scaled production, called the "Together Again" tour.

This demand required a compact, portable sound system, set up for touring. Hank Cattaneo, production supervisor for the show, knew what was required.

"When this tour was first announced, we had more 200 calls from production-related companies...the sound, lighting, trucking and staging people," Cattaneo says. "Everybody wanted to get in a bid, because this would be a prestigious event. Many sound companies tried to sell on the basis of their equipment alone—speaker brands and so forth.

"But, for this show, people are more important than the equipment. I expect the

David Scheirman is RE/P's live sound consulting editor and is president of Concert Sound Consultants. Julian. CA.

sound system to be well thought-out already, and to function as it is designed. The personnel that come with the system were the critical part, as far as we are concerned."

Cattaneo and Alan Richardson, sound-mixer for the Sinatra/Martin/Davis show, examined a wide variety of proposed systems and narrowed the choice down to four major touring sound suppliers. A-1 Audio, Hollywood, CA, and Las Vegas, NV, was selected to provide and set up a self-contained, portable and modular sound system with 360° coverage, to be suspended from the ceilings of the tour's different venues.

System preparation

After reviewing the proposed itinerary and arenas the tour would visit, Alan Richardson and A-1 Audio selected a loudspeaker system comprising 24 Meyer MSL-3 enclosures. The trapezoidal cabinets would be hung from only four points, with each group of six enclosures suspended from a 1-ton hoist. Sound system technician Dan Kasting supervised the assem-

bly and preparation of the system in A-1's Hollywood shop.

"Today, the more quickly a touring sound system can be set up, and the less hassle it is to use in a flexible manner, the more interesting it becomes to production managers," Kasting says. "We are not the only people to fly speaker systems for shows in the round. There are several other shows on the road right now that do the same thing. It's become a regular technique for some road shows. But, we've worked hard to come up with a hanging system that goes together, gets in the air, comes down and packs up quickly and easily."

Pre-tour preparation included the fabrication of special wiring harnesses so that cable runs could follow the lighting trusses for a clean look. Rigging gear was assembled, and a wide variety of extra bridles, shackles and hardware fittings was gathered. Groups of speaker enclosures were suspended from ceiling beams at the A-1 shop, and the static load figures (combined total array weight when not in motion) were recorded. All wiring lengths, sound equipment inventories and other system parameters were computer-documented.

Hanging speaker system setup

All of the system's cable, rigging and chain-motor boxes are equipped with heavy-duty truck casters. Inset metal-protected spaces are provided on top of each case that correspond to the casters, so the cases can be readily stacked on top of each other. Venue-supplied forklifts are used to unstack the boxes prior to their being rolled to the stage area in the center of the arena. (See Figure 1.)

The loudspeaker enclosures travel in groups of three with removable metal dollies underneath each cabinet. Protective wooden covers are provided for each speaker cabinet, and are attached with Velcro-type straps.

The speaker enclosures are held in place for truck travel by a unique, custom-built metal hanging bar with swivel points. The same compact, lightweight hanging frame that keeps the three enclosures together as one unit for rolling across arena floors and for cross-country truck travel is also the device used to attach hanging hardware to the cabinets. (See Figure 2.)

The two-point metal swivel bar is connected to the loudspeaker enclosures by inserting clevis pins into the speaker boxes' integral pan fittings. Each three-part frame assembly can then be set to tailor the horizontal coverage of the three-box array as needed to suit particular daily setups. Steel aircraft-cable safety loops are run between every separate hardware part, thus providing the needed extra pro-

Figure 1. Equipment cases have sturdy lid inserts to seat the casters of other cases when stacking. Venue-supplied forklift is used to unstack the load.

Figure 2. Custom metal hanging frames, with swivel points for array angling, are used to hold together each group of three boxes. Wire rope slings are used to pick up the load.

Figure 3. As an added feature, steel aircraft cable safety links are used to secure all hardware parts.
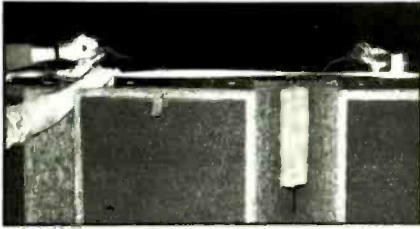
Figure 4. Closed-cell foam wedges are placed between adjacent speaker cabinets and used as spacers to insure optimum horizontal coverage angles.
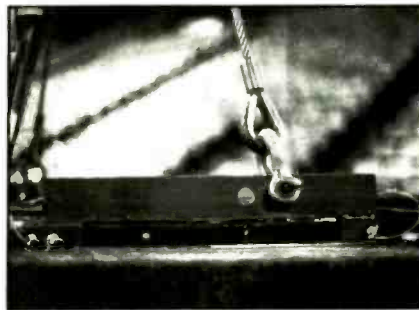

Figure 6. Custom compact metal hanging frames are used on the top and bottom of each speaker array.

Figure 5. Rigger uses screw-pin anchor shackles to attach steel bridles to lifting line prior to hoisting.

Figure 7. Downward tilt of the lower row of enclosures is determined by choice of hole or wire rope sling attachment.

Figure 8. Six Meyer MSL-3 enclosures hanging in place from a single motor. Note rigger atop lighting truss.

tection against falling items should a bolt, weld or structural joint fail in service. (See Figure 3.)

Wedges, or "stoppers," made of closed-cell foam, have been crafted in various widths, and are color-matched to the speaker enclosures' carpeted gray exterior finish. Once the desired horizontal coverage has been selected for a particular ar-ray, the foam wedges are held in place with a ratchet-strap that takes up hard-ware system slack and tightens the array in preparation for hoisting. (See Figure 4.)

The loudspeaker system is suspended from points located on ceiling structural beams. Riggers go aloft with lifting ropes, or "drag lines," to bring items up into the ceiling area. First to go up are beam wraps, or "softeners," often old gunny sacks that can be wrapped around struc-tural beams to help protect the wire-rope bridles and to keep the bridles from slip-ping while under load. Forged, load-rated shackles are used to attach the wire-rope bridles to the lifting lines and to the load. (See Figure 5.)

All rigging gear and motor-control lines to serve one hanging array point, and a 1-ton C/M-Lodestar motorized chain hoist, are contained in one of A-1 Audio's cus-tom rigging trunks. Once the rigger has pulled the loose chain up to the ceiling and attached it to the bridled hanging point, electrical power is supplied to the motor controller and is run up off the ground to head height and prepared to re-ceive the loudspeaker array load.

As the first group of speaker enclosures is hoisted into the air, signal cables are connected, all rigging fittings are checked and the next set of three enclosures is rolled underneath. Wire rope slings are available in lengths of 1½ feet, 2½ feet, 5 feet and 10 feet for setting the spacing between the upper and lower rows of en-closures. (See Figure 6.) A metal hanging bar with two different drilled attachment points is hooked onto the lower row of en-closures. The downward slant or angle of the lower three enclosures is adjusted by hooking into one of these holes with a screw-pin anchor shackle, which is con-nected to the wire rope thimble. (See Figure 7.)

Each single hanging point is used to sus-pend an array of six 278-pound speaker enclosures. The total weight of the array, including motor and rigging, is approxi-mately 1,870 pounds. (See Figure 8.)

While the rigging cases include a varie-ty of polyester or synthetic fiber slings (SpanSet products are typical), Dan Kast-ing prefers to work with wire rope slings whenever possible.

"I can't bite through steel," Kasting says. "The nylon slings are very handy for quick fixes, and we can do a lot with them in terms of securing cables and so forth, but they should never be used in permanent installations, because when they do get fa-tigued or damaged, it is not as obvious as on wire rope slings."

Kasting finds that suspending a loud-speaker system for performances with in-the-round seating can be challenging.
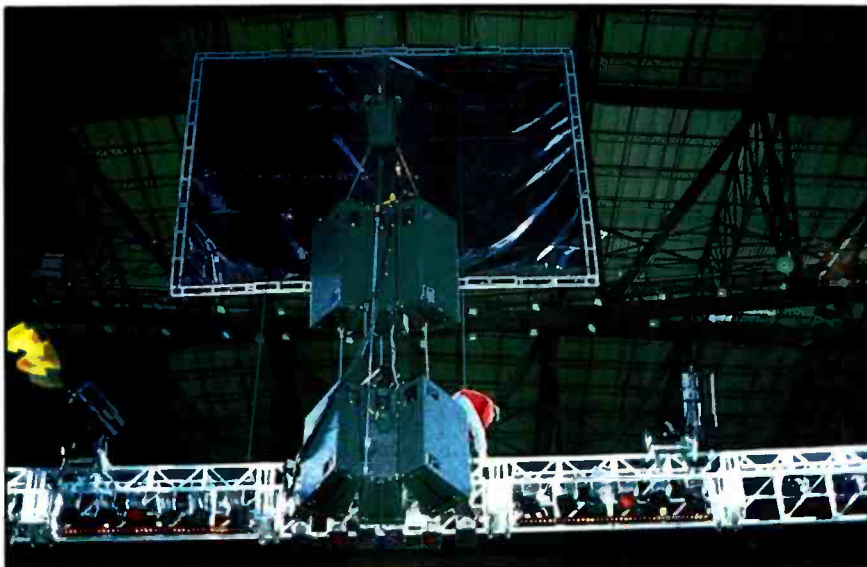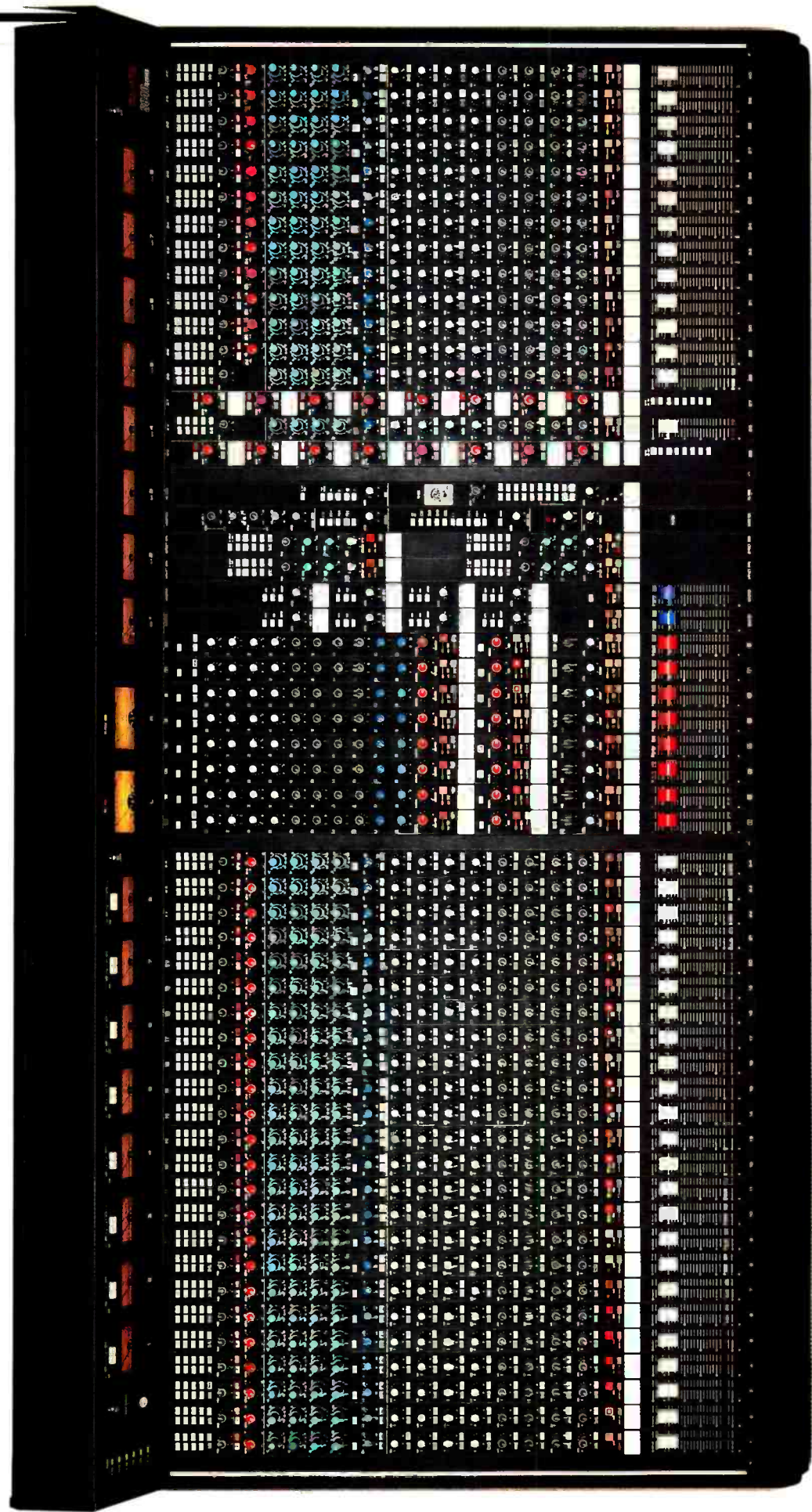
Our house... is a very very very very very fine house.
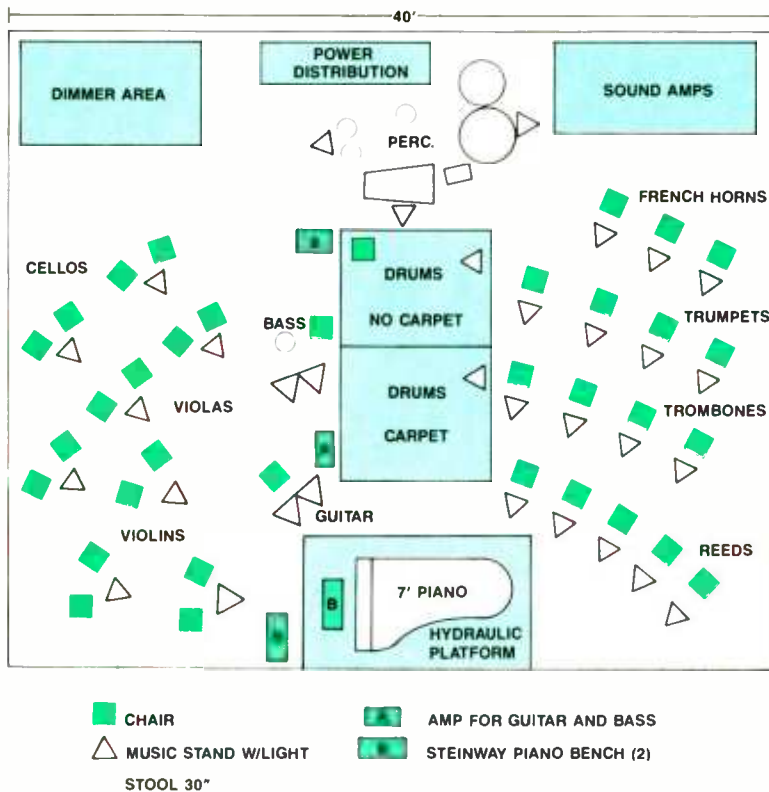
Figure 9. Orchestra pit/equipment location diagram.

Figure 10. Power amplifier racks are loaded with Crest model 4000s.

Figure 11. Four Meyer UPA-1A angled speaker enclosures serve as stage area monitors for the vocalists.

Figure 12. Shure's new wireless microphone transmitter for hand-held use with SM87 head capsule. (Units are resting on Yamaha PM-3000-40 mixing console.)

"It can be simpler, from a rigging perspective, because the hanging points are usually symmetrical," he says. "Typically, we are set up in the center of the arena floor instead of at one end, so we don't have to deal with the oval shape of some ceilings. At the same time, we are pointing the speaker system in four different directions instead of one, so instead of a single reflecting surface, usually the back wall in a normal setup, we have four walls to deal with. The array height and the horizontal and downthrow coverage angles then become critical factors."

Stage setup

For the "Together Again" tour, a 24-foot square stage area is centered on the floor of each arena. Only the featured vocalists, singly and together, set foot on the stage platform. All other musicians and orchestra members are positioned to one side of the center stage. This 40-foot area usually extends right to the side wall of an arena floor. (See Figure 9.)

All lighting system dimmer equipment, ac power distribution panels and loudspeaker amplifier racks are enclosed within this area along with the orchestra. Four of A-1 Audio's custom amplifier racks, each loaded with four Crest model 4000 power amps, are available to drive the loudspeaker system. A separate Cannon Australia EP4 audio connector on the back panel is dedicated to each Meyer loudspeaker cabinet in the hanging arrays. (See Figure 10.)

A typical rack also includes both a Meyer M3 system processor and a B2A processor for Meyer subwoofers (unused for the "Together Again" tour). With 24 MSL-3s in use, a total of 12 Crest 4000s are employed, with a 30A electrical circuit available for each pair of amplifiers.

As a standby technique, every amplifier rack is equipped with a wiring harness that terminates in extra XLR connectors. A single processor in one rack can drive the entire speaker system with the quick addition of a few patch cables, if necessary.

Stage monitor requirements are relatively simple. A total of four Meyer UPA-1A loudspeakers, resting on their sides and angled up, cover the vocalists' stage area with a single mix. (See Figure 11.)

A single Meyer UM-1 slant monitor is supplied to Sammy Davis Jr.'s drummer, and five small powered Anchor speakers are available for other areas in the orchestra pit, including the conductor's stand. Monitor levels are set at the house mixing position.

The 4-piece rhythm section and 33-piece orchestra (including string, brass, reed and percussion sections) travel with the show to each city on the tour.

"Because we have the same players every night, we are able to get a smooth

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sound with consistent mic placement techniques," says A-1 Audio technician Jim Stark. Flutes, clarinets and saxophones use Sennheiser 421s, as do the trombones. French horns are given Sennheiser 409s, while trumpets receive Shure SM-58s. An SM-58 also picks up the single acoustic bass violin.

Cellos, violas and violins are each given a Countryman Isomax miniature microphone.

"Every instrument gets one," says A-1 Audio technician Connie Fernstrom. "We like to clip them face down towards the instrument body, attached near the bridge."

A selection of Shure, AKG and Sennheiser microphones are placed around the drum sets and percussion instruments. A C-Ducer tape-strip pickup is mounted on the acoustic piano.

Vocal microphones in use for the show include some of Shure's new W15HT wireless units with SM87 head capsules. Equipped with a rubberized non-slip hand grip, the lightweight units have proved to be very reliable. (See Figure 12.)

"We carry a total of eight," says Alan Richardson, house soundmixer. "It's a brand-new unit that uses a diversity antenna system. It seems to have a very low noise floor. We need at least three working units during the show, plus a spare, and with eight available frequencies we've had pretty good luck so far."

With four different loudspeaker arrays, each facing a different direction, plus foldback speakers, downthrow and frontfill systems to adjust, Alan Richardson and the A-1 crew use a specific speaker system check-out technique each day.

"I like to start with the cabinets on top," Richardson says. "We turn up the speakers with the longest throw distance first, and set the corresponding processors at the amp racks to full gain. Then, speakers with shorter throws are brought in, and the processors are backed off for those arrays to give even coverage characteristics to all seating areas."

Four special radial horns are suspend-

Hanging system do's and don'ts

Touring sound companies and permanent installation contractors regularly hang speaker systems in auditoriums and arenas around the world. The hardware and mechanical techniques that have evolved to serve this industry have come into being to serve specialized needs.

A typical arena tour will travel in self-contained fashion, with multiple chain motors and all necessary rigging gear to get the equipment up in the air and keep it there. Most sound system rental companies own and maintain at least some bridles, slings and shackles for suspending loudspeakers. Many major firms own their own chain motor hoists, while others rely on specialized rigging firms to handle the job.

While every sound company uses a slightly different method to suspend loudspeaker systems, most professional riggers have developed safe, reliable and consistent ways to insure that the heavy gear goes up safely, and stays in place. Here are a few tips:
• Always use the right sling, shackle or bridle for the job. (Don't be tempted to substitute incorrect parts.)
• Purchase or rent the highest-quality rigging hardware that is available. (Don't try to get by with unrated bargain-store bolts, wire rope or slings.)

• Have any custom-built hanging frames or grids safety-checked and certified by a licensed structural engineer. (Don't let your Uncle Harry tool it up with the drill press in his basement.)
• Always use safety hooks and positive-contact latches or shackles when hoisting anything overhead, even a simple line. (Don't rely on duct tape or overhand knots. Even a nylon rope can injure someone when dropped from 50 feet or more.)
• Use adequate padding or "softeners" on beams and other metal-contact points. (Don't risk cutting and kinking slings due to improper preparation.)
• Know the safe load ratings of every piece of equipment in the hanging system, and allow an adequate safety margin. (Don't guess at speaker array weights, and don't exceed hardware load ratings.)
• Always sling your loads properly, using identical hardware parts when possible and rigging the load in a symmetrical fashion. (Don't use a single sling to suspend a two-point load. You risk slippage on the hook or shackle.)
• Rely on qualified riggers to locate and fix your hanging points. (Don't send the new guy on the crew up in the ceiling just because he says

he likes mountain climbing.)
• Always check motors and hanging hardware daily before it is put to use. Replace any worn components immediately. (Never assume that just because it came down OK last night, it will go up again today.)
• Always discard damaged rigging components. (Don't attempt to save money by hammering bent shackles or stitching ripped slings.)
• Always take the time to be safe. (Don't rush a rigging job.)

The use of common sense and some basic engineering principles that have evolved to serve the construction and mechanical trades will result in suspended loudspeaker systems that are safe and secure. There are no magic formulas or shortcuts when it comes to suspending loads in the air, and the time taken to become familiar with proven, correct rigging techniques is well spent.

A pocket-sized, "Handbook for Riggers," is available from Newberry Investments, P.O. Box 2999, Calgary, Alberta, Canada T2P 2M7. This 120-page booklet contains information on wire and synthetic ropes, hardware items, load attachment techniques and more.

Attachments to loudspeakers

Bolts, shackles, clips and eye bolts all develop the greatest strength along their axis—vertical orientation in hanging applications. It follows that the safest location for hanging attachment points will be at the tops of cabinets to minimize angular stresses on hardware.

This requires that the cabinet be strong enough to be safely hung from its top. Where multiple enclosures are needed, this can result in cabinets hanging from other cabinets. This configuration makes the loudspeaker enclosure an integral part of the hanging hardware system.

A 5:1 design factor is generally assumed for hanging hardware. It follows that loudspeaker cabinets must be capable of similar design factors. Few loudspeaker systems and components are load-rated and suitable for hanging without modification. The secure attachment of hanging hardware is no assurance that a cabinet will not pull apart under its own weight. An unmodified cabinet will be no stronger than the material that it is made from and the joinery techniques used to assemble it. *As a general rule, all wood and wood-fiber loudspeaker systems of more than 50 pounds require structural reinforcement for hanging installation.* There are many different methods of reinforcing cabinets that can provide acceptable safety margins, two of which are shown in Figure B.

For plywood enclosures, hanging hardware is shown bolted to steel reinforcement plates that are securely attached to the cabinet in a steel-wood-steel sandwich configuration. One corner is shown. All load-bearing panel intersections should be similarly reinforced with steel plates. This reinforcement method is not suitable for wood fiber or particle board cabinets.

Particle board and wood fiber cabinets should be externally reinforced with continuous steel strap or welded steel channel secured to the box so as to surround the enclosure completely, capturing dadoed-in baffle and back panels. This reinforcement method is suitable for all types of cabinets. If the baffle board isn't dadoed into the side walls, the cabinet shouldn't be hung and an appropriate substitute should be found. Never rely upon the internal bond strength of particle board or wood fiber cabinets to carry the weight of a large (more than 50 pound) system.

Small loudspeaker systems are subject to the same mounting considerations. Because they are small and fairly light, however, installers tend to make assumptions that frequently prove unsafe in the long run. Although the structural failure of a small loudspeaker cabinet seldom presents a serious hazard, it can be avoided by anticipating conditions that could affect the choice of mounting techniques.

Caution: Small loudspeaker systems often employ snap-in grill assemblies, which usually attach to the cabinet with Velcro or similar reusable fasteners. These mounting techniques, while satisfactory for home use, should not be relied upon for overhead installation in public places. Appropriate modifications are required.

When particle board cabinets are to be suspended from T-nuts and eye bolts, installers should be aware of loading limits that attend this practice. New particle board will exhibit an internal bond strength of 60-70 psi (ASTM D-1037). A ¼-20 T-nut in ¾-inch material will subtend approximately 1.4 square inches of bonded surface, resulting in a nominal (breaking) strength of 85-98 pounds. Using an assumed design factor of 5, the maximum axial load on a single T-nut would become 17-20 pounds. Reduce these factors by one-third for half-inch material. This is for particle board that is new or in new condition only. Clearly, this is not an acceptable suspension method for large loudspeaker systems.
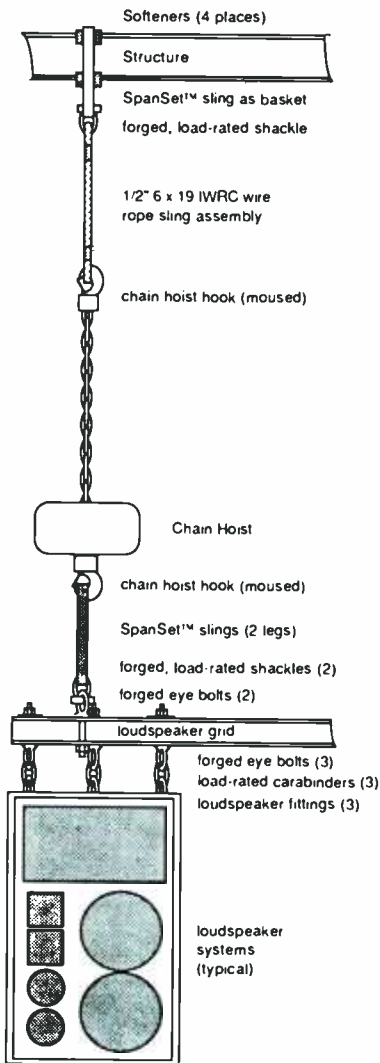
Figure A. *Typical rigging chain.*

Softeners (4 places)
Structure
SpanSet™ sling as basket
forged, load-rated shackle
1/2" 6 x 19 IWRC wire rope sling assembly
chain hoist hook (moused)
Chain Hoist
chain hoist hook (moused)
SpanSet™ slings (2 legs)
forged, load-rated shackles (2)
forged eye bolts (2)
loudspeaker grid
forged eye bolts (3)
load-rated carabinders (3)
loudspeaker fittings (3)
loudspeaker systems (typical)

Figure B. *Cabinet reinforcement.*

Steel Plate
Lifting Eye
cabinet
Steel Corner Bracket
"Sandwich" Construction (Plywood Only)

Steel Girdle Construction
Bolt to cabinet 6 8 inch spacing, typical 4 sides
Cabinet
Lifting eyes in vertical members
Weld (4 Places)

How to read a Kurzweil spec sheet.
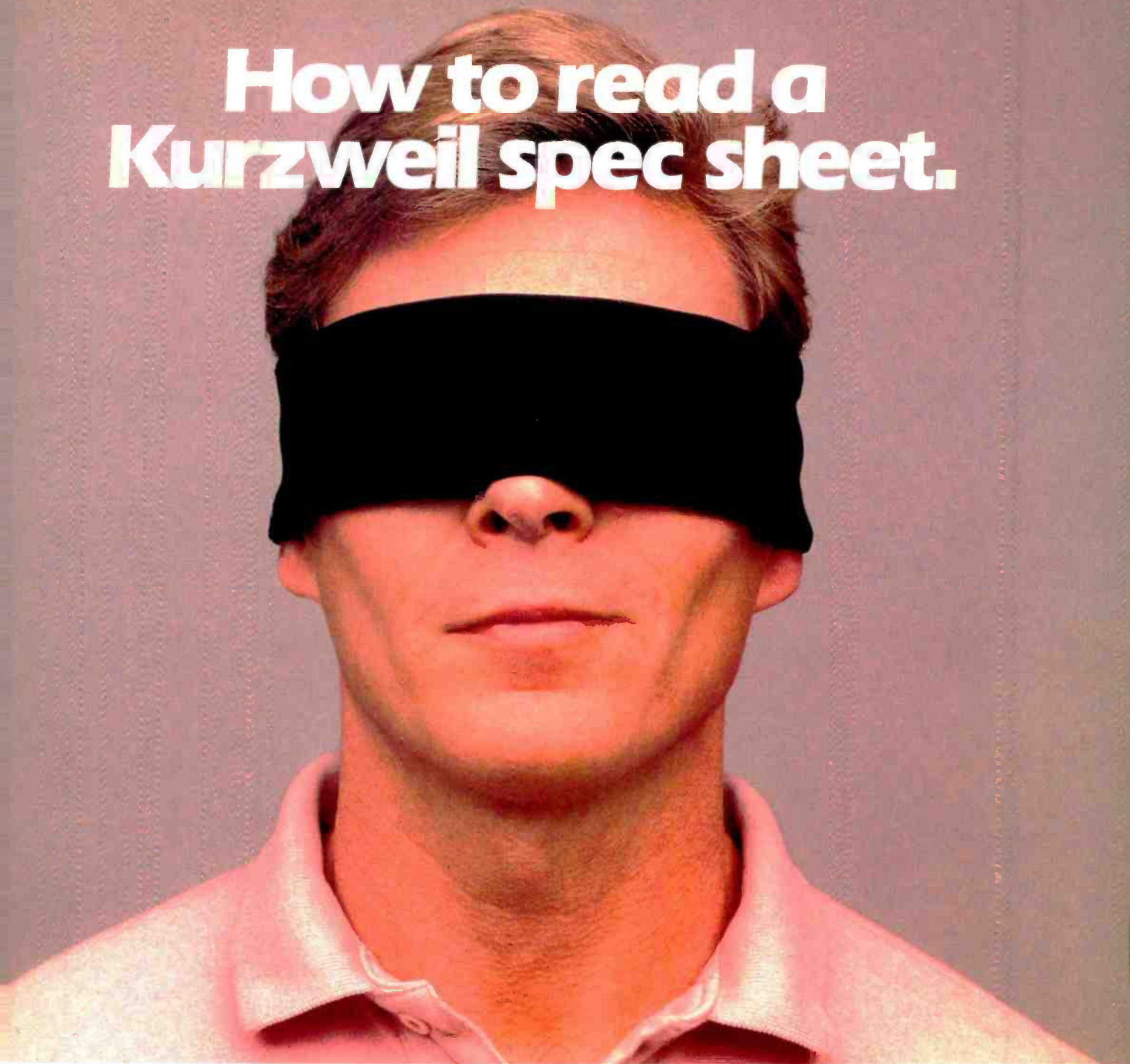
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ed from the lighting trusses, and are pointed down into "dead" coverage areas at the four corners of the stage area. A total of six Meyer UM-1 enclosures are mounted on wooden boxes and serve as a frontfill system. These house speaker subsystems are adjusted from the matrix outputs of the Yamaha PM-3000-40 mixing console. A single matrix output supplies a mono feed to the 360° coverage MSL-3 array.

Mixing the show

"This is a very vocal-oriented show, as you might imagine," Richardson says. "Those recognizable voices are what the audience came to hear. I have plenty of gain available on the orchestra. The average level of a typical show is well below the feedback threshold of the string section mics, but we can't afford to bury those voices, or to let the show levels become offensive.

"With ticket prices at $40 or so, I feel it is important that the folks get to hear this music the way they remember it, and that includes understanding the words to their favorite songs."

Richardson found the lower frequencies to be particularly critical on this show. "There just cannot be any boominess. It just kills the sound of this produc-tion. We go to great pains to get rid of a lot of the bass resonances that might be considered acceptable on a more rock-oriented concert."

Showtime

Regardless of one's taste in musical styles, listening to Ol' Blue Eyes and his buddies in the round is definitely an impressive experience. Dynamics of the show were subtly controlled by the conductor and his orchestra, creating a lush backdrop for the individual voices. Tunes such as Sinatra's "Mack the Knife," or Davis' "Candyman" brought appreciative ovations from the sold-out audiences in arenas such as Cleveland's Richfield Coliseum and Cincinnati's Riverfront Coliseum. Even in such cavernous, reverberant spaces as these, the combination of a finely adjusted 360° sound system and the painstaking attention to equalization and soundmixing techniques carried the crooners' words and music to the farthest, upper seating areas with surprising clarity.

This simple, compact hanging speaker system would lend itself well to expansion. The addition of available subwoofers and more amplifier/speaker array modules would easily accommodate rock-oriented performances.

"It's really a piece of cake, as far as hanging systems go," Kasting says. "When it's time to call it a night, this gear is down from the ceiling, all wrapped up and out at the trucks in about an hour and 10 minutes on the average."

The attention to system design details in such areas as cable and connector choices, custom hanging hardware components and loudspeaker packaging can mean the difference between a suspended sound system that saves time and makes money, or one that is a headache to stage crews, sound technicians and production managers alike. Designed to put "sound in the round" up in the air and get it down again as efficiently as possible, this system from A-1 Audio is representative of today's trend toward integrated, versatile hanging systems for touring use.

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Acoustical Measurement Techniques for Sound System Equalization

By Ralph Jones and John Meyer

This article examines the basic criteria for acoustical instrumentation, surveys the evolution of acoustic measurement techniques, and offers comments on future directions for the field.

Virtually every audio professional is familiar with the basic idea of measuring and equalizing loudspeaker systems. From studio to concert hall, the practice is widespread, and it is generally acknowledged that carefully considered adjustments to a loudspeaker system's response can almost always improve both sound quality and intelligibility.

Accordingly, market demand has stimulated a proliferation of sound system measurement and equalization tools. Audio professionals today face a sometimes bewildering set of choices that involve not simply brands of hardware but also conflicting opinions about what to measure and how to measure it.

Acoustical data

To quantify the performance of an acoustical system with the highest degree of accuracy, a measurement instrument must reveal three things about the system: (1) the frequency response; (2) the phase response; and (3) the coherence factor of the measurements. The ability to gather all three of these is essential in professional acoustical testing; ignoring any one of them involves significant compromises that must be carefully weighed.

Most audio professionals are familiar with the concept of frequency response, and few would question its importance. However, we are used to thinking of the phase-domain characteristics of a system as separate and less important than the frequency response. (In fact, the question of whether phase non-linearities are audi-

Ralph Jones is a Los Angeles-based, free-lance technical writer, engineer/producer and a regular contributor to *RE/P*. John Meyer is president of Meyer Sound Laboratories, Berkeley, CA, and a Fellow of the AES.

ble is still debated in the audio world.) Yet, when we speak of frequency response and phase response, we are really talking about two complementary representations of a single reality. Both are necessary to form a complete picture.

To understand the practical importance of gathering phase data, consider a simple "thought experiment." Suppose that a loudspeaker system preserved all of the amplitude relationships by frequency of its input signal (flat frequency response) but reversed the time order of the signal (like playing a tape backward). Certainly the system would be considered useless for any practical application, yet to an instrument that gathered and displayed only amplitude data, it would appear perfectly acceptable. A simple frequency response plot obviously would present an incomplete and misleading picture of this system's behavior.

We can supplement this illustration with another that was first proposed by the late Richard Heyser. Imagine a 2-way loudspeaker system consisting of a low-frequency element at 1 meter from the measurement point and a high-frequency element a mile farther away. If the high-frequency element were sufficiently powerful to overcome propagation losses, it would be possible to equalize this system for apparent "flat" frequency response at the measurement position, but its time response would be rather bizarre. Again, an instrument that did not gather phase data would be incapable of revealing the obvious sonic inadequacy of this system.

The concept of coherence factor, which is integral to FFT (Fast Fourier Transform) analysis, is least familiar to the audio world—although it is of fundamental im-

portance. The coherence factor is an expression of the degree of linear relationship between the test signal and the signal that is gathered at the point of measurement (the system output signal).

Coherence is diminished by any contamination of the test signal through the object of measurement, which for our present purpose is the loudspeaker system and its environment. Such contamination may include noise at the system input or output (electronic glitches or ambient sounds), frequency-domain distortion products (from over-excursion, coil rubbing, cabinet and diaphragm resonances, and clipping), and phase-domain distortions (uncompensated time delays from resonances, driver offsets, acoustical reflections and so on).

It is easy to see that all of these factors affect the validity of measurement data to some degree. Harmonic distortion products, for example, show up as added high-frequency energy that, in an amplitude-only display, would be interpreted as an indication of better high-frequency response than is actually the case. Because that high-frequency energy is not present in the input signal, but is created by distortion in the system itself, it does not respond to equalization. Unless the test instrument provides some method for excluding such effects, there can be little confidence in the data, and the measurements will be of questionable usefulness.

Measurement resolution

Beyond the type of data gathered, another fundamental criterion to be considered is the frequency resolution of the instrument.

The audio community disagrees on pre-

cisely what constitutes appropriate resolution for acoustic measurements. There are those who argue, for example, that instruments that reveal phenomena spanning less than a third of an octave "show more than we need to know." The implication here is that relatively narrowband frequency response variations in loudspeaker systems have little audible effect, and, by extension, correcting them will yield little audible improvement.

A moment's reflection is sufficient to discover that, if this point of view is valid, it must be for the special case of speech reproduction. Consider that our Western-tempered scale divides the octave into 12 steps, and the music of many other cultures employs far smaller microtonal divisions. It follows that a loudspeaker response peak or dip, spanning one-twelfth of an octave or less, can affect not only the sound quality of music but also the balance of pitches.

This was demonstrated strikingly by the National Bureau of Standards' tests of the CBS Copycode system for R-DAT recorders, which showed that the extremely narrow notch employed in that system was capable of eliminating notes in passages of instrumental music. It would be difficult to argue that such effects are insignificant, much less that they are inaudible. After all, the NBS proved that the Copycode notch was demonstrably audible to a statistically significant population of experimental subjects. This was one of the main reasons that the proposed Copycode system was ultimately rejected.

The cumulative experience of many audio professionals over the past decade indicates that very fine frequency resolution is essential in critical acoustical measurements. To form a truly accurate picture of a system's response, we know that 1Hz resolution is desired at low and mid frequencies; at high frequencies (10kHz and above), 50Hz resolution will suffice. Instruments that display measurement data only at resolutions lower than this again impose a compromise that carries real consequences. Among these are that the instrument inevitably obscures such narrowband phenomena as resonances in high-frequency driver diaphragms, midrange

FFT in concert

A live concert is very dynamic and the acoustics of the venue can change dramatically. Conventional techniques for adjusting EQ use clicks, chirps or pink noise as the test signal; these are obtrusive sounds and cannot be used when the audience is present.

The room, however, changes acoustically when filled with people; sound distribution, reverberation times and floor reflections change. The humidity and temperature variations that the audience and the stage lighting cause alter the sound absorption characteristics of the air. [See "The Speed of Sound" by Dennis Bohn on page 38 in the April issue—Ed.] The net result of all these are shifts in room resonances, which can ruin the EQ.

In general, speaker room response is measured using a ⅓-octave analyzer, and the observed resonance peaks and notches are then equalized with ⅓- or ⅙-octave graphic equalizers. Unfortunately, many low-frequency rumble and midrange ringing problems can be caused by narrowband resonances that are too narrow to be measured in a ⅓-octave analysis. Nor can they be corrected by fixed-bandwidth graphic equalizers.

The use of live music as the test signal, adjustable equalizers and an analyzer capable of pinpointing narrowband resonances would be the ideal solution. This would mean that the EQ could be adjusted for changes in the room acoustics when the audience arrives and corrected for further changes during the performance.

An FFT analyzer, used in conjunction with high-quality parametric equalizers, provides the answer.

A dual-channel FFT analyzer is designed to investigate causal relationships in any system in which both the cause and effect are electrical signals, or can be converted to electrical signals by transducers such as microphones.

The analyzer samples and averages the data in both signal inputs and, by comparing the two frequency spectra, computes the frequency response and other characteristics of the system. Because this type of analyzer operates on both signals and uses the system input as the reference for analysis, the technique is independent of the test sign. Therefore, any test signal, including music, can be used.

Using an FFT

An FFT analyzer is easily connected to sound reinforcement sys-
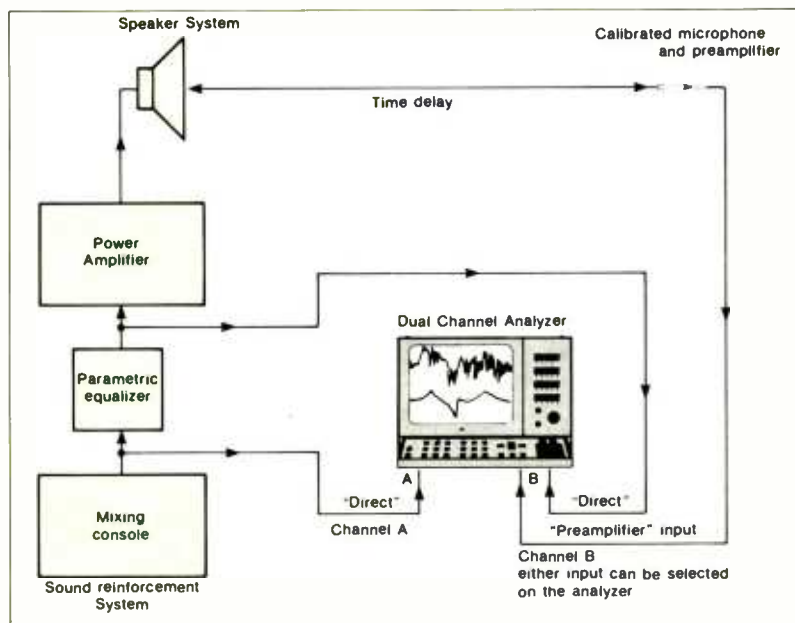
Figure A. Block diagram of a dual-channel analyzer connected to the sound system. Measuring between the A and B inputs signals, with music as the test signal, the analyzer computes the room response.

acoustical resonances and anti-resonances in the space, and so on—all of which audibly alter the reproduced sound.

Time selectivity

It is imperative that any measurement instrument for acoustical analysis employ some form of selective time windowing. To understand why requires some background theory.

The off-axis frequency response characteristic of every practical loudspeaker system differs to some degree from the on-axis. Directional control is achieved relatively easily at high frequencies, for which the wavelengths are generally quite short relative to the size of the corresponding driver. At low frequencies, for which the wavelengths become greater than the

driver and cabinet size, loudspeakers become omnidirectional.

For this reason, low frequencies tend to predominate (relative to higher frequencies) in the energy with which a loudspeaker excites the reverberant field of a space. As a consequence, the steady-state frequency response of an installed loudspeaker system as measured at a typical audience location exhibits a falling characteristic with increasing frequency, because it includes both the system's direct energy and the predominantly low-frequency reverberant energy in the space.

Early attempts at testing installed loudspeaker systems employed real-time analyzers with pink noise excitation to measure the steady-state response. Research-

ers used ⅓-octave equalization to achieve a measured flat response characteristic. Subsequent research in human psychoacoustics, however, has proved this approach to be inadequate.

It has been shown that the ear perceives the frequency response of an installed loudspeaker as a combination of the system's direct energy and the earliest reverberant reflections. The time period encompassing the direct sound and early reflections is called the Haas window, after the researcher who first described the phenomenon. Although the length of the Haas window varies slightly depending upon the timing and strength of early reflections, it is, in all cases, far shorter than

---

tems to measure room response. (See Figure A.) You need two signals for the measurements. The system input for channel A is the output of the mixing console. The channel B input is from one of two sources in turn: the output from a calibrated microphone located in the audience (usually in the house mix position) and the output of the equalizer.

An initial frequency response measurement has to be made between the console output and the calibrated microphone, with live or recorded music as the test signal. From this measurement, you can calculate the time delay between the

speaker system and the microphone, which enables you to set the time delay into the analyzer so that the channel A data is delayed until the microphone response reaches channel B. The built-in, user-defined delay eliminates the need for an external delay line and can be used to time-align the speaker system.
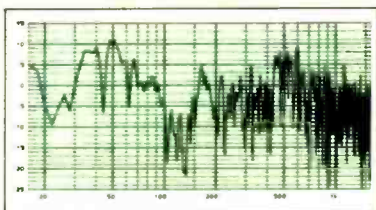
A frequency response measurement without EQ then gives you the speaker room response. (See Figure B.) This measurement can be stored and displayed while, with the delay switched off, the inverse frequency response of the equalizer is continuously measured and displayed in

real time on the split screen. This enables you to set up the EQ (See Figures C and D.) to compensate for the speaker room response and to produce the overall response you want. A further speaker room response measurement, using the microphone and time delay, with EQ applied shows the effect rendered through equalization. (See Figure E.)

In performance

While the audience is arriving, you can check the EQ using recorded music as the test signal. Just before the concert starts, the EQ will have been adjusted for a full house. The FFT enables you to continue monitoring the speaker room response during the performance, using the live music for measurements. You can continuously correct the EQ as acoustic conditions change.

Audience reaction and other random noises tend to spoil the measurements, but here a coherence function on the analyzer can help you. Coherence is a measure of how well the data in channel B correlates to channel A. Looking at the coherence while measuring, you can see data averaging reduces the effect of random noise. Continue measuring until you are sure that you have taken enough averages for good coherence and, therefore, a valid response measurement.
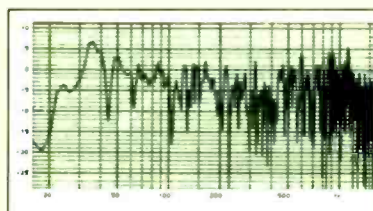

Figure B. Room response measurement without equalization.
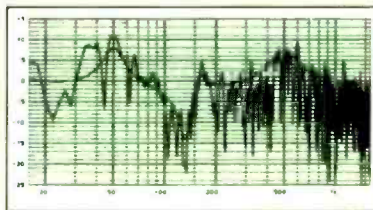
Figure D. Figures B and C combined to show the fit between the room response and the equalizer.
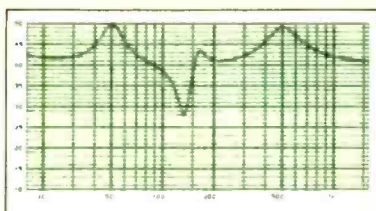
Figure C. Inverse frequency response measurement of the equalizer.

Figure E. Room response equalization with equalization.

Guide to Comparing High-Power Amplifiers

By Pat Quilter

The ideal power amp combines "old fashioned" install-and-forget reliability with modern standards of performance and improved headroom.

Live sound systems have continued an impressive evolution in quality despite a lack of major technological breakthroughs since the late 1960s. At that time, you were lucky if you could actually hear all the musicians on stage. Modern touring systems can fill arenas and outdoor amphitheaters with well-mixed, high-level sound that is not only physically exciting but clean enough to impress an audience that has been raised on good home stereo systems.

Much of this success is due to the improvements in power amplifiers. Compared to the grand-daddy of modern amps, the Crown DC-300, today's amps can deliver three to four times the power in less rack space and with similar weight. Reliability has also improved, but is still critical.

In touring systems, amplifiers must perform for a season with very few breakdowns. With this many amplifiers, cost is also a major consideration. Power amplifier components carry a great deal of stress, and expensive overdesign is undesirable from both cost and weight standpoints. The ideal combines "old fashioned" install-and-forget reliability with modern standards of performance and greatly improved headroom. This requires a clear-headed review of the available technology, thorough attention to design details and construction, and advanced protection and control systems to ensure the improved performance is safely and effectively delivered.

Don't focus on a single performance parameter. A chain is only as strong as its weakest link. An excellent amplifier must have equally good circuitry, mechanical

Pat Quilter is vice president of engineering for QSC Audio. Costa Mesa, CA.

and thermal design, input/output interface, controls, displays, documentation and appearance. Be wary of claims of only one "correct" circuit or approach; many well-respected amplifiers have succeeded using various output devices and circuits. Success is determined by the overall quality of engineering and construction, as well as by the manufacturer's backup service. The goal is to secure the highest possible performance with the lowest lifetime installed cost, which is not necessarily related to the in-the-carton price.

Selecting for performance

In many cases, amplifier selection gets bogged down in the issue of which amplifier "sounds better." To test for this, it is essential to adjust both amplifiers for equal volume, as people will judge the louder amplifier to have more "presence," even when the volume difference is not obvious. With output levels matched, you will find that "glaring differences" suddenly become virtually indistinguishable.

Most reasonably modern amplifier designs have perfectly acceptable noise, distortion and frequency response. There are small differences in tonality between amplifiers, but most evidence indicates that these are primarily "spectral" variations, such as small deviations in frequency response. Amps that look flat on the bench, using resistive loads, may display small frequency response errors into actual speakers. High frequency damping factor is rarely checked, which could cause perceptible "harshness" or "sweetness" in high-frequency drivers.

Because speakers have uneven impedance vs. frequency curves, if there is any coupling between speaker line currents and the input signal, there could be a frequency-dependent feedback, especial-

ly with unbalanced inputs. So, balanced inputs are preferred. Small differences in high- or low-frequency response also may be audible to well-trained ears.

The point being: The frequency response of the program source and speakers is subject to much greater variations. Dramatic claims that an amplifier has (or lacks) "resolution," "depth" or "harshness" will be put in perspective if you substitute "a small difference in frequency response" for any of the above expressions. Obvious problems in actual use will usually be traced to a lack of headroom (clipping) into real-world loads, false triggering of protection circuits, poor input/output connections, and, of course, faulty operation or shutdowns for any reason, including breakdown.

Amplifier output headroom

Headroom is the amount of undistorted peak power available. Amplifier power is the product of peak voltage and peak current. It costs money to increase either variable. High-voltage transistors cost more, and high-output current requires more or larger transistors. There will be a certain load impedance at which both the voltage and current peak out together; this is the impedance into which the amp delivers its maximum power.

For instance, say an amplifier can deliver a maximum of 40V and 10A. If we connect an 8Ω load at 40V, 5A of current will flow. Multiplying these two numbers gives 200W, which is the amplifier's power into an 8Ω load.

However, not all of the current capacity is being used. If we connect a 4Ω load, the same 40V will cause 10A to flow, which equals 400W. We are now getting the full power of the amplifier. If we connect a 2Ω load, 40V would cause 20A to

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AMPLIFIER A

200W — 8 OHMS

330W — 4 OHMS

AMPLIFIER B

200W — 8 OHMS

200W — 4 OHMS

AMPLIFIER C

200W — 8 OHMS

200W — 4 OHMS

CLIPPING
8—OHMS

CLIPPING
4—OHMS

HIGH CURRENT DESIGN

4—OHM CLIPPING SLIGHTLY
LOWER DUE TO LOADING
BUT NORMAL IN SHAPE.

CURRENT LIMITING

4—OHM CLIPPING MUCH
LOWER AND HARSHER
DUE TO CURRENT LIMITING.

V—I (FOLDBACK) LIMITING

SEVERE SPIKE DISTORTION
CAUSED BY INTERACTION
WITH REACTIVE LOAD

FIG 1

flow, which is greater than the amplifier's current limit. The limit is 10A, which will result in only 20V across a 2Ω load.

Not only has the power rating dropped back to 200W, but, because heat is related to current flow, this loading is far more stressful on the amplifier than the 8Ω load. This simplified example illustrates that all amplifiers have an optimum load impedance that is determined by the amplifier designer.

With real-world speakers and amplifiers, the situation is more complicated. Speakers and passive crossovers (if any) have impedances that vary at different frequencies. A speaker that averages 8Ω may have an actual impedance that ranges from 4Ω to 16Ω. An amplifier that is optimized for maximum power into 8Ω will not have the required current to drive the speaker at frequencies in which its impedance dips to 4Ω, and the output will distort prematurely. It may also distort in abnormal ways. (See Figure 1.)

To avoid these problems, look for an extra margin of power at an impedance 50% less than your load impedance. To get real-world 4Ω performance from an amplifier with a "400W, 4Ω" rating, it should show a rating of about 600W into 2Ω. If the 2Ω rating is 400W or less, this indicates inadequate protection-circuit limits or a lack of

output current for full performance into real-world 4Ω loads. Because there are many 4Ω speaker systems, especially subwoofers, this can be a primary cause of headroom differences between supposedly identical amplifiers.

This also explains why "high-current" designs are desirable. However, you also need high-output voltage to prevent clipping where the speaker impedance rises. Because there are a lot of 4Ω and 8Ω drivers, most professional amps have maximum power ratings at 2Ω to 4Ω.

Output devices and circuits

The original high-power transistor was the so-called "bipolar" type, which required special protection circuits for reliable operation. Bipolar transistors have no inherent current limit, so if the load impedance is reduced (or shorted out), currents can increase to destructive levels. "Current limiting" or even "current foldback" circuits were devised that limited or reduced the current into abnormally low-impedance loads. These circuits can have unwanted effects on clipping, as shown in Figure 1.

Modern bipolar devices are much more rugged, and about 10 years ago, we started seeing amplifiers with very high current limits that are cut back (or shut down) only

after serious, prolonged over-current operation. This permits full-peak output voltage into normal speaker impedance dips.

Also during this period, another type of power transistor, the MOSFET, became available. Early supporters claimed awesome benefits, including more "transparent" sound, simplified circuits and self-protection. A level-headed review of the facts is necessary to appreciate the tradeoffs. MOSFETs have inherent current limits, which reduce automatically as the temperature rises. This makes them more immune to self-destruction into low impedances, but also reduces peak current capability into normal impedance dips.

Claims that there is "no current limit" are false. There may not be any current limiting circuitry, but the devices themselves have current limits that cannot be controlled by the designer. They also have somewhat less output efficiency—more power is lost in the devices as waste heat.

MOSFETs are touted for simplified circuit design, and it is true that some bipolar amplifiers are more complex. However, it is possible to streamline the bipolar circuit, achieving self-protection and simplicity comparable to MOSFET designs, while having more voltage and current output for a given size of power supply and heat sinking. This translates into more

usable output power for a given cost and temperature rise.

Another popular claim for MOSFET designs is greater inherent fidelity, or "transparency." In fact, amplifier fidelity is primarily determined by overall circuit design and board layout. The output devices are certainly important, but they are only a part of the overall circuit.

MOSFETs are said to avoid thermal runaway. This claim is based on the same self-limiting characteristic, but only low-efficiency MOSFETs have stable bias points and effective thermal compensation circuits. Because FETs have higher temperature rise under heavy loads because of their reduced efficiency, the quality of the heat sink design rather than the type of output device becomes the major thermal question.

Power FET research and development is focused on single-polarity (n-channel) parts for use in switching power supplies. These parts are not well suited for conventional linear amplifier design, requiring more complex "quasi-complementary" circuits and bias compensation. The most attractive application would be for high-frequency switching amplifiers, or "Class D" circuits. After being "just over the horizon" for at least a decade, this type of design is finally being produced.

Class-D performance trade-offs

"Linear" amplifiers represent the most direct approach for amplifying analog signals, and have been continually improved since 1915. All amplifiers start with a (more or less) fixed power supply with matched positive and negative voltages. Positive and negative "valves" (vacuum tubes or transistors) are arranged in a "push-pull" circuit. which controls the amount of positive and negative power available to produce the desired speaker motion. The maximum output level is reached when either valve is turned fully on; beyond this, the amplifier "clips."

Actual devices can't quite pass 100% of the power supply voltage; the residual loss is high for tubes, moderately high for FETs and fairly low for bipolar transistors, which is one reason bipolar output devices are popular.

Regardless of output devices, however, all linear circuits have inherently high losses at typical output powers such as the FTC's infamous "⅓ power" test. This is because linear "valves" act as variable resistors, which take in the full voltage of the power supply, dissipate the unwanted portion as heat and send the desired output level to the speaker. Losses are fairly low in the fully on or off condition, but very high at about half output.

Because all intermediate linear output levels are inherently lousy, the only way around this problem is to operate the output devices as switches where they are either on or off. To reproduce the complete range of output levels, the percentage of on and off time is varied from zero to 100%.

Obviously, this has to be done at a rate that is much higher than the audio frequency, so that the on-off pulses can be smoothed out; for good audio quality at 20kHz, it is necessary to switch at a rate of 500kHz. This requires much faster devices than linear circuits, and it is extremely challenging to duplicate the audio quality we are used to getting in linear designs. In particular, the circuit needs a large output filter to smooth the 500kHz pulses, which reduces damping factor, especially at high frequencies.

A much simpler improvement that dramatically raises the efficiency of the linear output circuit is to add one or more extra power supply levels, which divide the full

An alternative viewpoint

By Frank Kelly

Editor's note: Because there are several design approaches to power amplifiers, various manufacturers were sent an outline of the summary points of the main article and were invited to respond. The following response outlines UREI's philosophy and responds to some points made in the main article.

1. Don't focus on a signal performance parameter or feature.

Because there are more similarities than differences found in today's professional power amplifiers, users may have to decide upon a single feature that's critical to their application. These choices often include items such as UL listing, amplifier size and cooling method, and the reputation of the manufacturer.

2. Improving amplifier output headroom.

Amplifier headroom is often misunderstood by users and subject to misrepresentation by manufacturers. Because most modern amplifiers have at least 100dB of dynamic range (10dB better than a compact disc), headroom should not be a problem if the system levels are properly adjusted. If the situation occasionally demands greater output, it is better to have an amplifier that clips cleanly than one with a poorly regulated power supply designed to yield more "output headroom." The clean-clipping amplifier will produce more useful output power with processed audio signals. It will also run cooler than the increased headroom unit and have less power supply-induced ac ripple on the output waveform at the clip point.

3. MOSFET devices—balancing the trade-offs.

The decision to use MOSFETs in power amplifiers is strictly a marketing one. Technically, the choices are clear. MOSFETs do not sound like tubes or significantly self-protect themselves, to list a few of the common misconceptions. Besides, an amplifier's sound character and reliability depend far more on other

Frank Kelly is chief engineer at UREI, Northridge, CA.

factors than the output transformers. Further, MOSFETs cost more, are not available in well-matched complementary parts and have not been proved to provide a clear advantage to the user.

4. Reliability.

Product reliability, quality and consistency are probably the most important factors to consider when choosing an amplifier. What good is the latest featherlight hi-tech amp if it sounds bad, or worse yet, shuts down? This is a significant reason why amplifiers use traditional "iron horse" technology. Professionals who make their living with these tools cannot afford to take chances with their business.

5. Cooling.

There are two ways to cool an amplifier: natural convection cooling without a fan and fan-assisted cooling. Convection cooling usually requires a larger heatsink than fan cooling. Therefore, amplifiers cooled by natural convection are usually a rack space or two larger, and a bit heavier than their fan-cooled counterparts. On the other hand, fan-cooled units usually attract dust and need to be cleaned periodically. All fan-cooled amplifiers produce some noise, which can be a problem in studio control rooms. Finally, like anything having moving parts, fan reliability should be considered. This can be a major consideration in large sound systems having many amplifiers.

6. Interface.

Many amplifier manufacturers offer accessories in an effort to provide system approach to their customer's needs. Amplifiers are available with built-in crossover filters, attenuators, optional transformer inputs, stereo/mono switches that save extra wiring, output transformers and autoformers. These accessories often save space and money. However, users should be sure that they are not compromising their system performance by using convenient amplifier accessories that may not match up to the performance of discrete, external components.

voltage supply into halves or thirds. With a reasonable amount of extra circuitry, the linear output devices can draw their power from the closest of these extra voltages, rather than having to "reach up" to the full power supply at moderate outputs. This greatly reduces waste heat without sacrificing the high performance of linear output circuits.

Reliability: success is no failure

Achieving improved amplifier reliability is accomplished by reducing parts count, using well-tested technology, replacing complex discrete circuits with high-quality integrated circuits (such as the multisourced 5532 op-amp, which was designed for demanding audio applications), improving the thermal design by high surface area, fan-cooled heat sinks and direct metal-to-metal mounting of output devices, and providing complete fault isolation of each channel (split or dual-mono power supplies).

A physically rugged design will not only qualify for road use, but will resist shipping and installation accidents. Long-term reliability in fixed installations is enhanced by convection cooling, gold-plated contacts in the signal path (all switches and connectors), proper derating of components and effective internal protection circuits that guard against the inevitable accidents and hazards.

Overall system reliability is improved by providing for gas-tight connections (barrier strips and binding posts), special control accessories and features for particular needs (plug-in accessory sockets).

Cooling: convection or fans?

Fan cooling is the lowest cost approach for packing more watts of power into a given rack space and allows some weight reduction as well. This makes it the obvious choice for portable systems, as long as the amplifier designer has provided adequate fan capacity for high-level operation. The primary disadvantages are noise and dust build-up.

Most installers of fixed audio systems prefer convection-cooled amps because the heat sink surfaces are large and require only gentle air flow for full cooling. Dust build-up, if any, is more gradual; more dust can be tolerated before maintenance, and dust is externally removable. If the rack is closed, a fan may be needed to maintain airflow. However, a single fan with a large external filter is easier to maintain than individual internal fans, and its failure need not result in amplifier overheating for reasonable duty cycles.
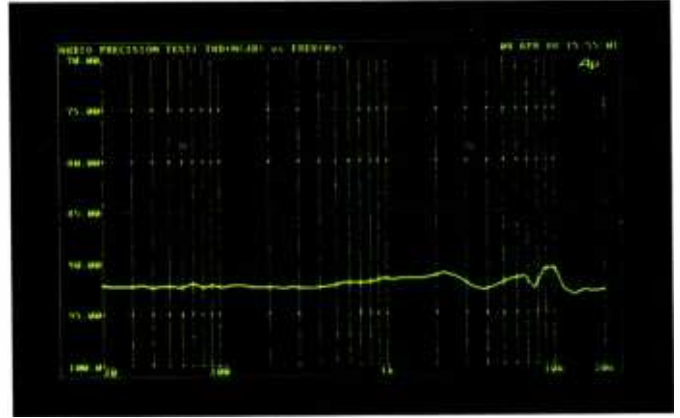
Some touring companies used to prefer convection-cooled amplifiers because they could increase cooling by several hundred percent, if needed, with external fans. To-

DUAL—MONO POWER SUPPLY

```
CH A                        CH A
POWER      —[FUSE]—         DC POWER   —[ AMP CH A ]
TRANSFORMER                 SUPPLY

CH B                        CH B
POWER      —[FUSE]—         DC POWER   —[ AMP CH B ]
TRANSFORMER                 SUPPLY
```

SPLIT POWER SUPPLY

```
                  —[FUSE]—   CH A
SPLIT                        DC POWER   —[ AMP CH A ]
POWER                        SUPPLY
TRANSFORMER
                  —[FUSE]—   CH B
                             DC POWER   —[ AMP CH B ]
                             SUPPLY
```

COMMON POWER SUPPLY

```
POWER       —[FUSE]—    MASTER        —[ AMP CH A ]
TRANSFORMER             DC POWER
                        SUPPLY        —[ AMP CH B ]
```
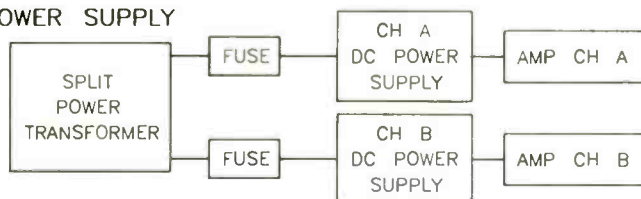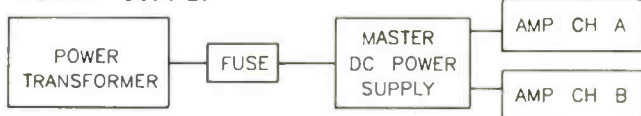
FIG 2

day, however, there are a number of proven designs with internal fans.

Interface

Even with the best design and construction, an amplifier can be useless if not properly designed for professional installations. A well-designed amplifier facilitates good installation workmanship and system monitoring by providing suitable connectors, protective features and circuits, and diagnostic displays, such as level readouts, clip indicators and thermal warning lights.

The cheap and ubiquitous ¼-inch connector has gradually given way to the more costly XLR connector in professional systems. In a stationary installation, or where an umbilical cable connects to the rack, a barrier strip provides the cheapest gas-tight connection. Balanced inputs should be standard in order to reject cable-induced noise; ground loops and interference pickup often are the single most objectionable problem in otherwise good systems.

Ac switches and circuit breakers should be mounted on the front for convenient power control and reset. Gain controls can be front-mounted for convenience, if a provision is made for lock-out covers when needed. Otherwise, the rear location is the lowest-cost location that prevents tampering. All panel features should be recessed for protection against damage.

Complete protection circuits are essential to reliability, but they must not interfere with audio performance. Some amplifiers rely on shutdown circuits or fuses blowing to protect the output devices, which involve interruptions and manual reset. Another undesirable form of short circuit protection is called "VI limiting," or "current foldback," which can misbehave with reactive speaker loads and cause a gross "snapping" distortion, as shown in Figure 1. A better form of protection permits high currents to flow for short periods (normal audio peaks) but cuts back smoothly into gross overloads.

If an output device shorts in a direct-coupled amplifier, the power supply will be dumped into the speaker, and it will be destroyed. This situation calls for "dc fault" protection for input as well as output faults, an important feature that still is not standard on all brands. Turn-on and turn-off muting eliminates a common distraction; for full protection, muting should operate immediately if ac power is lost for any reason, not just when turning the amp off with its own switch.

Displays can facilitate system debugging and servicing by providing essential diagnostic information. Indicators may be included that show power on/off, protection circuit activation, output signal, clipping and excessive temperature. Of course, if the amps are not mounted in a visible location, these features may not be worth the extra cost.

Fail-safe installations

While reliability is important in all professional applications, it is absolutely critical in some. This requires the installer to consider the ugly question: "What if something breaks?" System redundancy and fault isolation become necessary defenses.

Full-fault isolation in power amplifiers requires "mono design." Each channel must have its own power supply, fuse or breaker, and control and protection circuits. However, because of size and limited demand, few high-quality single-channel amplifiers are built for professional service. Fortunately, a few manufacturers are building "dual-monaural" designs. This term means that two channels are built in a single chassis for reduced size and cost, but with fully independent components, as described above.

A "split power supply" provides most of the same benefits by using a single power transformer with dual secondaries feeding separately fused power supply circuits. The lowest-cost approach of a common power supply for both channels increases the chance that a bad channel will impair or force shutdown of the remaining channel. (See Figure 2.)

For continued operation after a failure, the load must be driven by two or more channels in parallel or a provision made for switching the load to a reserve power amplifier. Normally, direct-coupled high-performance amplifiers cannot be paralleled into a single load without gross instability.

This problem is solved by at least one amplifier brand that offers "auto backup" in the bridged-mono mode. If either channel fails, its speaker protection relay will automatically ground its output, permitting the remaining channel to continue driving the load at 50% lower voltage and stress, ensuring continuity of performance until servicing. Most bridged-mono amps will lose the entire signal if either channel fails.

All things considered

Specifying and installing power amplifiers, like any other part of system design, is a demanding and exacting job. Even though it is possible to comb the marketplace and find an amp with the "perfect" power and features for each type of speaker, many system designers prefer to standardize on one or two models that cover all of their requirements.

Even though this practice may result in overkill in some cases, the time saved in market research, testing and service training, not to mention volume discounts, may offset the extra cost. In addition to the benefits of better-than-required performance and accumulation of a longer track record, system changes and upgrades are much easier.

Some amplifier manufacturers have realized this, and find that most of their sales volume is concentrated in a few well-equipped, well-tested designs that offer "the works" at competitive prices. ℞∕ℙ

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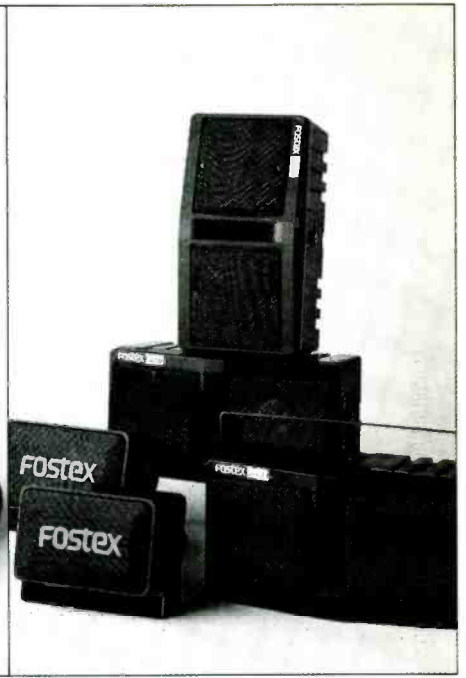
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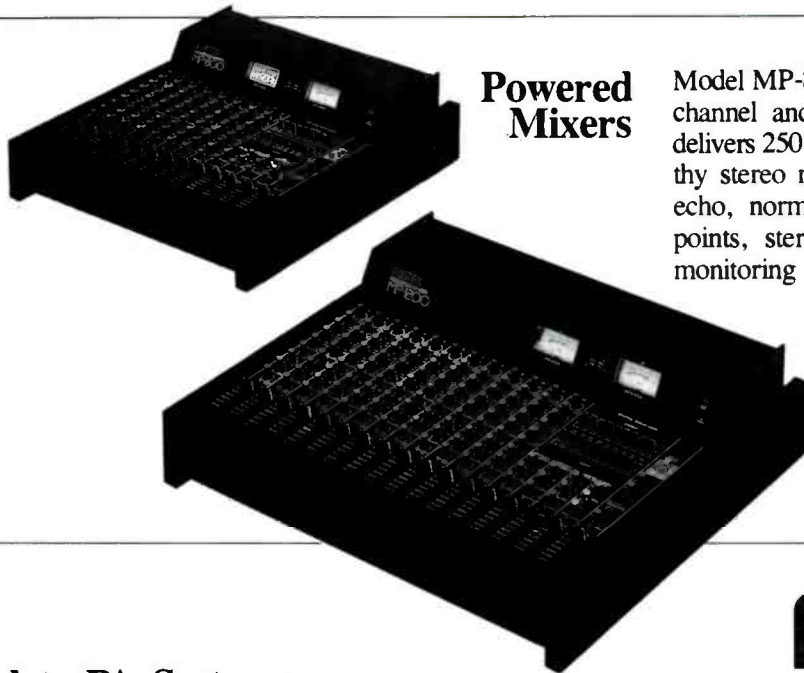
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Circle (25) on Rapid Facts Card

Active Crossover Considerations

By Terry Pennington

Several schools of thought exist concerning the theory and operation of active crossovers.

A bit of basic knowledge about the theory and operation of active crossovers is essential before you attempt to make a purchasing decision or begin the task of installing and operating any such device. There are several different schools of thought on the subject, and a dose of objective reasoning cannot hurt the process.

Consider the following facts:

1. All active filters (and, therefore, cross-

Terry Pennington is director of technical marketing and product development for Rane Corporation, Everett, WA.

overs) operating in the analog domain will exhibit a predictable amount of phase shift, a fact that becomes critical when one expects two adjacent drivers (low to mid, mid to high, etc.) to sum properly at a selected crossover frequency.

2. Steepness of crossover slopes is generally important to prevent damage to driver elements and to reduce the potential for signal distortion.

3. Input and output configurations and impedances must be evaluated to ensure that the component will fit properly with other components in the sound system.

4. As with all signal processing devices, care must be taken in the selection process to ensure that expected input levels at the crossover will not degrade the crossover's performance due to circuit overloads. Noise and distortion are related characteristics that tie well to the overload levels of the circuitry.

Phase shift and slopes

The major frequency of interest in a crossover is at the −3dB (−6dB in the Linkwitz Riley crossover) point when one is considering the phase properties of the component. At this frequency, the two adjacent drivers being driven (low to mid, mid to high...) must both reproduce the input signal equally. If there is a phase difference between the drivers, the level of the resultant acoustic energy will not be flat with respect to levels outside of the crossover area. Only an in-phase condition will ensure that the drivers will push or pull air in unison, resulting in flat acoustic energy through this narrow band of frequencies.

In the current state of the electronic arts, the only way to achieve an in-phase condition at the critical crossover frequency is through the use of an even number of "poles" in the crossover's filter. Each pole of the filter will exhibit a phase shift of 90° at crossover, so two poles gives us 180°, three equals 270° and four poles produces a phase shift of 360°. In this case, as in most others, 360° and 0° may be considered to be the same. A signal will be in phase with another at either 0° or 360°.

The alert reader will notice that two poles, an even number, yields 180° of
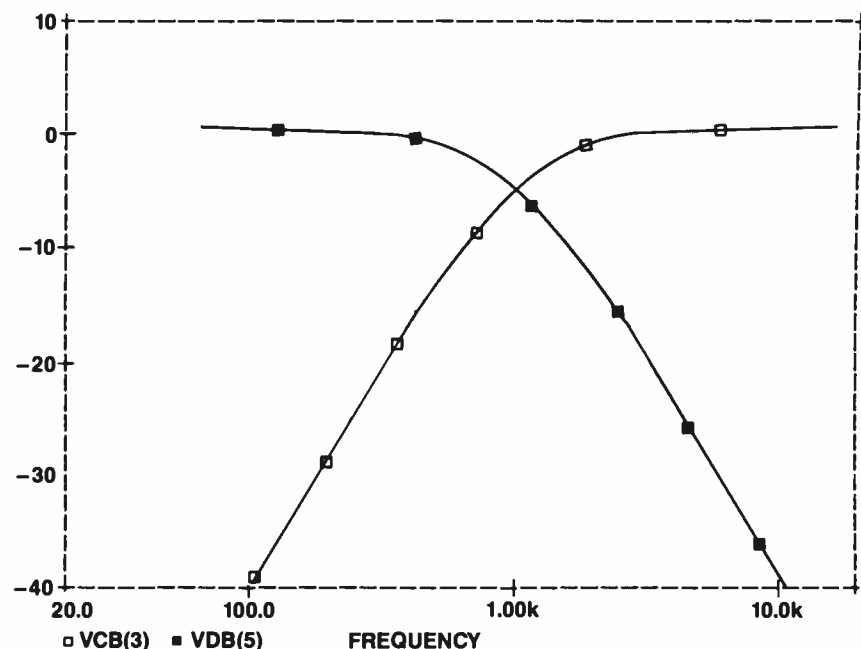
Figure 1. Second-order crossover.

□ VCB(3)   ■ VDB(5)   FREQUENCY

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phase shift. A signal inversion is necessary on one of the outputs to yield an "artificial" in-phase condition. The same would be true with a 6-pole, 10-pole or any other such even number that is not a multiple of four. Crossover filters are normally referred to either by their "order" or by their rolloff rate, not by their number of poles. The two are directly related. A 2-pole circuit (second-order) will exhibit a rolloff rate of 12dB per octave, a 3-pole 18dB per octave. Each pole or order will add 90° of phase shift at crossover and 6dB per octave in the rolloff specification.

A 4-pole (fourth-order) filter will generate the required in-phase condition between outputs without inversion. It rolls off at a 24dB-per-octave rate, which has an additional advantage. The steeper slopes relegate the actual crossover area (the frequencies at which the two adjacent drives must share information) to a smaller

window, further minimizing any phase-induced problems to a smaller and less audible range.

A very common crossover design continues to be the 3-pole (third-order), 18dB-per octave type. All crossovers of this sort will be 90° out of phase at crossover. If a crossover such as this is used, there will be some degree of acoustic loss at the crossover frequency unless other measures are used to correct the situation. There are ways of designing speaker cabinetry to minimize this loss, such that acoustic phasing compensates for electrical phase. (Sorcery is not a lost art.)

Maintaining proper phase relationships at the output of the crossover is not, unfortunately, the end of the story. Any misalignment of the drivers in the mechanical time plane can destroy the integrity of the crossover's otherwise fine job of preserving fidelity. Many fourth-order

crossovers available today offer a means of electrically compensating the output of the crossover to overcome misalignment of drivers. This is a feature well worth considering if your skill with a chain saw is below average.

Interface considerations

The majority of crossover products on the market offer balanced inputs and outputs as well as the option to run in the unbalanced mode. A typical sound system benefits from balanced operation, which offers greater immunity from noise induction than does an unbalanced interface. Balanced lines are not necessary, nor are they desirable in all cases, and should, therefore, not be mandatory.

A general rule of thumb regarding input and output impedances states that an input impedance should be as high as is practical and the output impedance as low
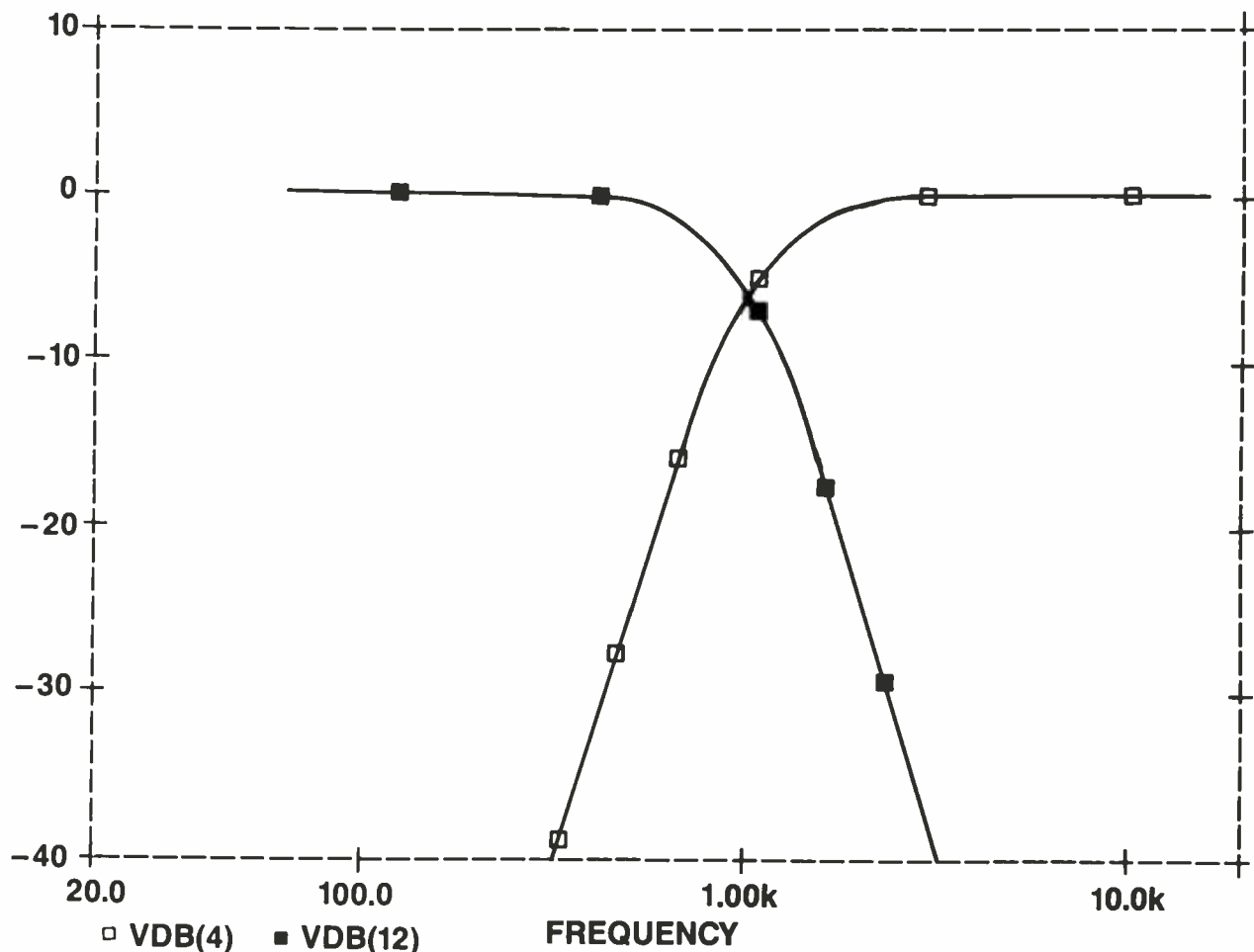
Figure 2. Fourth-order crossover.

as possible. A high-input impedance guarantees that virtually any component will be able to drive the unit under consideration without signal degradation or level loss; a low output impedance minimizes the amount of induced noise that will be picked up in the lines between the crossover and the power amplifiers. A low output impedance further guarantees that if the input impedance of the power amplifiers is unusually low, the signal loss between crossover and amplifiers will be minimized.

Signal levels and overloads

A basic requirement of all signal processors is that they accept necessary input signal levels and pass them through their circuitry without damaging the audio due to overload (clipping) or other forms of level-dependent distortion. Distortion created by internal overloads and speed limitations can seriously degrade the performance of the component under evaluation. Active filters, especially those required for the complex signal-splitting tasks of active crossovers, can produce subtle distortion if careful design considerations are not taken into account. By far, the most important aspect of this discussion is the internal gain levels inside the filter block. If these internal nodes are not carefully designed so that internal gains do not exceed the overall gain of the filter stage, some very bizarre waveforms may be generated, which will destroy an otherwise quality product.

Sharp filter angles, especially at high frequencies, require relatively fast amplifier stages (high slew rates). To ignore this requirement does not produce some of the subtle (possibly non-existent) slew-induced distortions that have been attributed to power amplifiers. This type of slew-induced distortion is measureable, audible and just plain obnoxious. Beware.

Conclusion

Selecting a crossover can be a simple task or one of great complexity and frustration. Be sure that the speaker manufacturer's recommended crossover frequencies are available on the unit you select, that the phase considerations of your system are taken into account and that your personal goals are met in overall fidelity. Having done all that, your system should benefit greatly from some of the fine crossovers available on today's market.

If you are considering your first purchase of an active crossover or simply looking to upgrade an existing crossover, you should check for the following points in all of the potential candidates:
1. What is the phase response between adjacent frequency bands?
2. If the phase response does not meet my

criteria, what has been done to ensure that the adjacent outputs will sum properly?
3. Will the unit provide enough gain for my application? (Crossovers normally do not provide a great deal of gain; if a lot of the amplification is required at the crossover stage, look elsewhere in the system for proper gain structure!)
4. Is the noise floor of the crossover low enough for the signal levels at which I wish to operate?

5. Are the non-linearities of the circuit adequate? (Is the distortion low enough?)

If you currently have an active crossover in you system, you may feel that its characteristics are degrading the sound. Check its data sheet for the above points. If the specifications mentioned above are not published, call the manufacturer. You have a right to know. **RE/P**

Introducing the
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Computers on the Road

By Scott Gershin

Planning, full backup and preventive maintenance are the keys to touring successfully with computers.

During the past few years, computers have taken major roles in the production of modern music, such as organizing synthesizer patches, creating musical timbres and automating musical sequences in conjunction with parameter and patch changes on outboard signal processing gear.

Because of the need to reproduce the sounds created in the studio, the computer quickly became an intricate part of the band on the road. Acts such as Journey, Madonna, the Pointer Sisters and Michael Jackson sequence their background vocals and horn and string arrangements by using a combination of samplers and computers.

Scott Gershin is a sound designer and effects editor at Todd-AO/Glen Glenn in Los Angeles and a free-lance writer.

Unfortunately, computers are not particularly roadworthy and have been known to crash, even in controlled environments. I interviewed operators and technicians from two well-known touring acts to learn how they ensure that their shows run smoothly and efficiently when depending so heavily on computers.

Michael Jackson tour system

I first spoke with keyboardist Chris Correll and his keyboard technician Mitch Macoulier with the Michael Jackson tour. Chris uses two Synclaviers, each having 64 voices with 32Mbytes of RAM, four 160Mbyte hard drives in a custom flight case, two super floppy disk drives, two Pericome monitors and eight tracks of Direct-to-Disk recording, all of which is fed into a 40-input Yamaha PM3000 console on stage and controlled by a Synth Axe. (See Figure 1.)

Mitch was retained as the tour's Synclavier technician. He previously worked for New England Digital as a hardware designer and technician. He was asked to design a setup that could withstand the day-to-day problems that commonly occur while on the road. After meeting with Chris and the Jackson management, Mitch designed a system that would take into account every problem they might face on a worldwide tour.

One of the most common reasons a computer crashes is an unstable supply of ac power. To deal with this problem, the tour carries its own power generators with a UPS (uninterruptable power supply) backup called "Show Power." In case of power failure, it can power the stage for 20 minutes.

The band gear, which includes the Synclavier and a variety of rack-mounted keyboards, uses eight 30A circuits that are ground-lifted at all points (except for the Synclaviers' central processing unit). This solves most of the grounding problems. Also, in line for all the racks and computers are Juice Goose PD-1s and PD-2s, which protect the equipment from spikes, surges or other dirty power conditions.

To combat the bump and grind of semi-trailers and constant setup and tear-down procedures, the road cases were built with durability in mind. All road cases were built to withstand worst-case conditions, and they feature heavy construction, extra shock-absorbing foam and the lowest possible center of gravity—especially for the hard-drive cases.

Preshow preparation

Before each show, Mitch opens the

Chris Correll, keyboardist with the Michael Jackson tour, with his Synclavier stage setup.

tower (CPU), vacuums the components, cleans the contacts and reseats each of the cards. By doing this, he ensures that all the cards are well seated and that nothing has worked loose in transit. Oxidation and dust contamination are also kept to a minimum.

After this, he runs a complete diagnostic check on both systems. If hardware problems occur while on the road, there is the equivalent of a third Synclavier in parts, assuring quick replacement either before or during the show.

Another precaution was the installation of a customized air conditioner filter, on the intake fans, to strain out such elements as smoke, dust and insects. This, too, helps keep the components and their contacts free from foreign particles that could cause the computer to malfunction at an inopportune moment.

During outdoor concerts (which make up a large portion of the tour), a specially designed tent structure is assembled under the stage to house the Synclavier towers, storage devices and most of the rack-mounted keyboards. Referred to as the "clean room," this tent is heavily air-conditioned and is under strict climate control. Computer-based systems are quite sensitive and can malfunction during sudden changes of temperature and humidity. The tent also controls excessive humidity changes that can cause oxidation—and adds to the overall safety of the equipment.

To avoid software malfunctions, disk erasures and lost sound files, each set of hard drives contains identical information, which can be downloaded into either or both Synclaviers. If the data on all four hard drives are lost, the show can be reloaded into the reformatted hard disks via a backup Kennedy tape drive.

Unexpected interference

While on the road, an unanticipated problem developed. During certain sections of the show and rehearsals, an intermittent, unknown source was triggering random notes on the Synclaviers. After searching many hours, Mitch and Chris narrowed the problem down to the Synth Axe. Apparently while Chris was facing in the direction of the nearby wireless communications equipment on stage, a signal was being picked up from the radios, which caused the Synth Axe to trigger random MIDI note on/off commands. A quick, steadfast rule was initiated that no radio communication equipment was allowed within a certain radius of the stage during performances. As an added precaution, the Synth Axe was lined with copper foil to prevent RF interference. (The Synth Axe company was notified and hardware updates are being designed to cancel any external radio signal that may cause interference.)

At one point in the tour, Mitch detected a cold-solder joint condition but could not pinpoint which connection was broken. He and the other technicians ended up resoldering all the connections in one of the Synclaviers. They felt that it was quicker and more efficient to resolder all of the connections than to search for the bad one.

The Pointer Sisters setup

To get another perspective on the potential problems of traveling with computer-based systems, I spoke with Greg Whelchel, keyboardist for the Pointer Sisters. Although their schedule is not as hectic
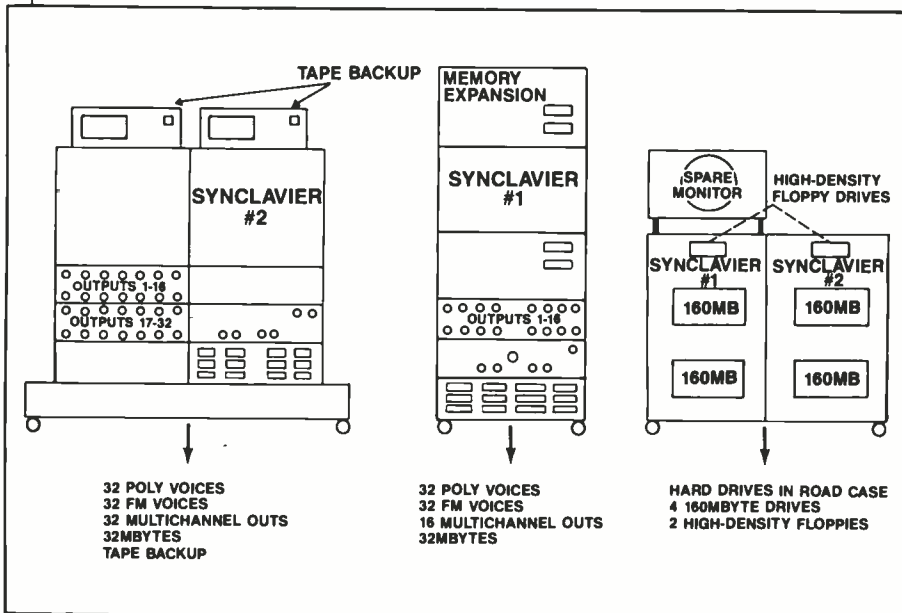
Figure 1. Off-stage racks used during the Michael Jackson tour.
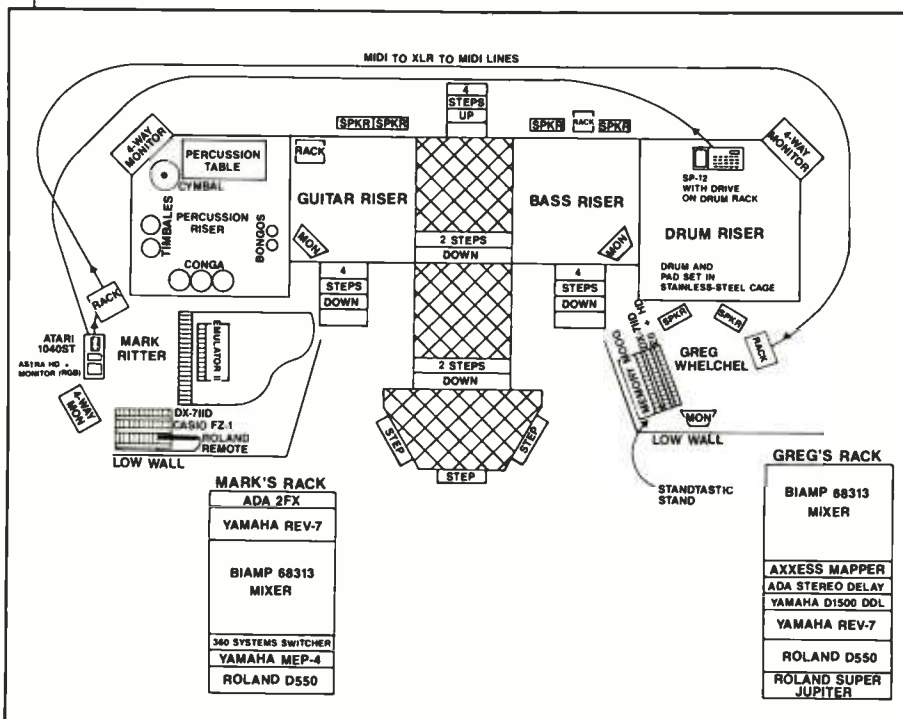
Figure 2. Stage plot for the Pointer Sisters tour.

as Michael Jackson's, they faced similar problems when trying to guarantee the consistency of each performance.

Greg uses an Atari ST computer with Hybrid Arts software, an Astra HD+ 20Mbyte hard disk and a color monitor. The computer is used to control patch changes and sequenced material. The sequencer is controlled by external sync from an E-mu SP12 drum machine, which the drummer controls. This allows the drummer to control the tempo of the show. (See Figure 2.)

A drawback in choosing the Atari computer is that it's not built to be taken on the road. He had to design a special road case so the Atari could withstand rough handling.

At all times during the tour, the computer and monitor live in their cases of 2-inch foam. One case houses the CPU and the Astra hard drive, and the other houses the color monitor. Because Mark Ritter (the second keyboardist in the band) and Greg don't have the luxury of their own keyboard technician, they do all the necessary connections to the computer and keyboard equipment. Their road crew only places the computer system at an appropriate place on stage. They developed this policy to ensure against potential setup and bootup problems.

In case of hardware problems, the band carries a second Atari system that is left in the wings during performances. It can be set up and implemented at a moment's notice.

To deal with power surges, dips and spikes, the computer and peripherals use Isobar line conditioners, which have been enough to do the job.

Every three weeks to a month, Greg and Mark go through their computers and keyboards and reseat all circuit cards, clean the contacts, and tighten all screws and bolts that might have come loose during travel.

The Ataris aren't very finicky when it comes to temperature, but during outside performances they are either moved out of the direct sunlight or covered with solar blankets.

Adapting MIDI cable lengths

During the design of their setup they noticed they had difficulties in establishing exact MIDI cable lengths. They realized the lengths would vary from house to house. Instead of interfacing their computer and keyboard equipment with 5-pin DIN plugs, Greg and Mark took advantage of the abundance of 3-pin XLR cables that the sound crew had, and made MIDI-to-XLR connectors that could be strung together to create any desired length.

Special note: MIDI currently accesses only the three inner pins of the 5-pin DIN plug, making it convenient to interface with XLR connectors. [See "Interface Design" in the January issue for more information regarding MIDI interfacing—Ed.] Also note that neither of the remaining pins (1 and 5) of the DIN plug should be connected to shield or ground. This can cause data errors, especially when in use with the Atari computers. Atari uses pins 1 and 5 for MIDI Thru information (which is not part of the MIDI standard).

The use of well-shielded mic cables also prevents added RF and other digital data from entering the data stream. Unfortunately, because of the frequency of digital information, the MIDI data could not be transmitted over microphone snake lines that were carrying audio information. To do so caused digital data to leak into the audio feed. The only solution was to wire each cable transmitting computer data directly to its destination.

Monitor troubles

Another unexpected problem occurred during the low-light and blackout portions of the show. The Atari color monitor defaults to a white screen with dark blue or black characters. The computer cast a white glow across the stage that the lighting designer hadn't anticipated.

To solve this problem, a utility program was used to change the colors of both the background and the characters/graphics. Greg and Mark used a dark blue background with yellow characters, thus eliminating the overcast of light.

Both bands had a problem when it came to the monitors. Because of the construction and susceptibility of the inner tubes to shakes and bumps, both the Pericom and Atari monitors had power supply problems that had to be fixed while on the road.

Possibly the point most emphasized by both acts is the importance of educating and training the road crew, other band members and management about the vulnerability of the equipment and the measures that must be taken to guarantee that the equipment works consistently and dependably. By training the road crew how to pack the equipment, and how fragile it is, by educating the management on the advantages of having trained specialists who can quickly and efficiently handle all problems that can arise during a long tour schedule, and by teaching other band members the system's setup and booting procedures in case problems arise when the operator or technician isn't available—these points are addressed.

Computers have become an integral part of our music making. Certain recurring themes become obvious when discussing the problems of touring with these delicate instruments. Learning a few basics and understanding the systems goes a long way in guaranteeing that each performance runs to everyone's expectations.

RE/P

Touring Checklist

It is important to foresee certain problems that can exist when touring with computers. The following checklist outlines items and elements:

Power requirements
- *Line conditioners.*
- *Surge suppressors.*
- *UPS units.*
- *House power vs. truck generator power.*
- *Circuits separate from lighting and house sound.*

Travel requirements
- *Road cases: Do they need casters? If so, what kind to absorb shocks and jolts?*
- *Internal shock absorption: Are foam inlaid cases necessary? How thick?*
- *Will it stack? Should it stack?*
- *What part of the vehicle has the least amount of jolts and movements?*

Backup systems
- *Is your show data backed up?*
- *If so, how hard is it to get to, and how long will it take to be back on-line?*
- *Do you need a second system off stage as a backup?*
- *Are you traveling with all the hardware components needed to get your system back on-line in the middle of the night?*
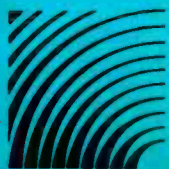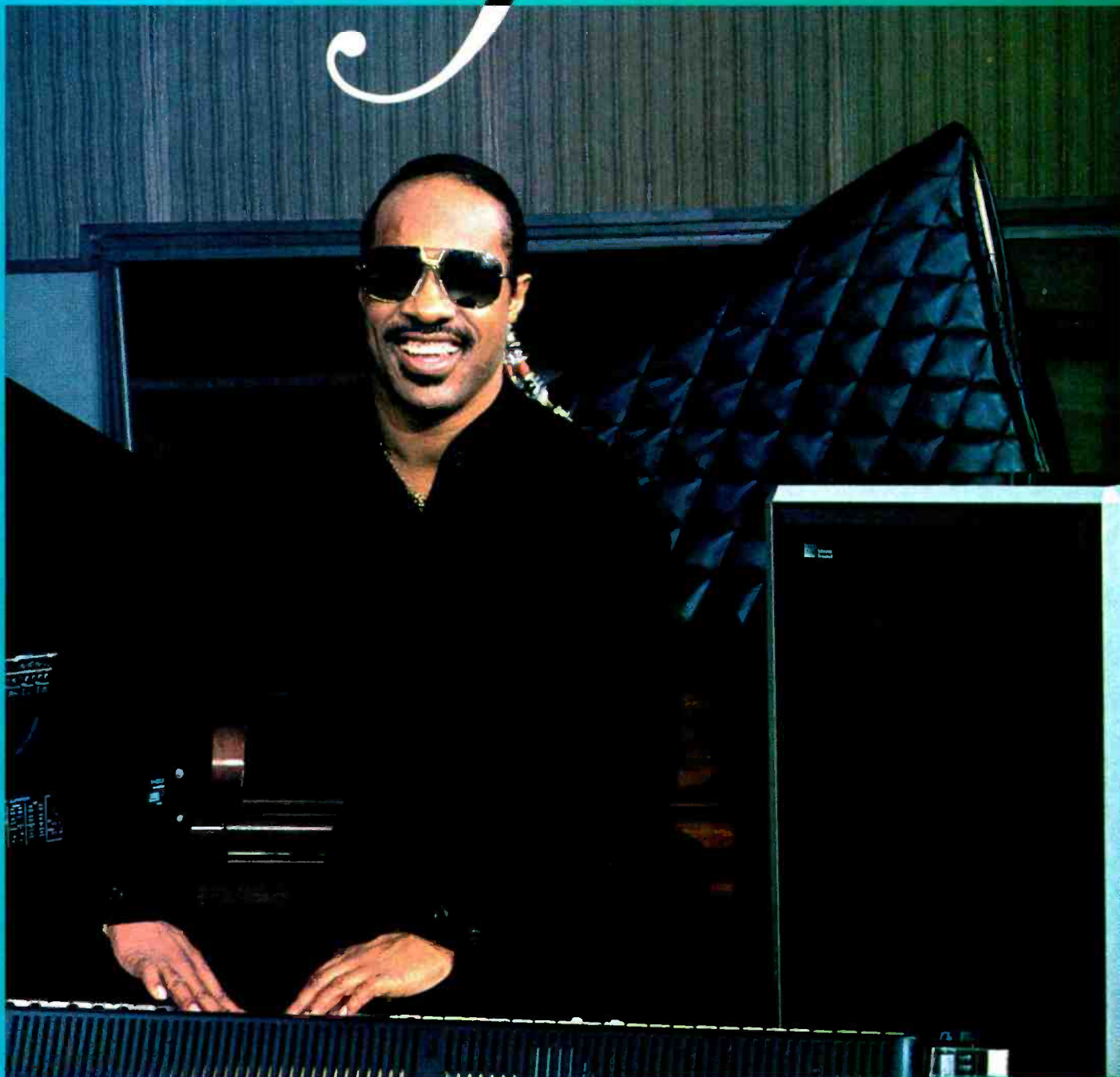
Personnel
- *Does your setup and tour schedule warrant the employment of a computer/keyboard specialist?*
- *Does the road crew know the do's and don'ts of the setup and handling of your equipment?*
- *What are your feelings on house crew members connecting your system, and does the road crew know those feelings?*

Engineer Interviews:

Robyn Gately and Paul Dalen

By Laurel Cash

Two viewpoints on the art of mixing sound for live performances.

Robyn Gately comes from a family of professional audio personnel. His father founded Gately Electronics, a much-heralded console manufacturer of an earlier era, and is now president of the David Hafler Company. Robyn has been involved one way or another in the audio biz since he was a toddler. Based in the Northeast, Robyn owns two audio companies: Joe's Sound & Salami, a manufacturer of speaker cabinets for the road, and Modular Sound Reinforcement, a respected sound reinforcement company that does mostly regional shows and some national tours.

RE/P: How did you get started mixing live sound?
RG: I had an interest throughout my childhood years because my father used to manufacture consoles for the audio industry. My first real job was a result of answering a classified ad in the *Village Voice* for a sound man for Bruce Springsteen. After working for Springsteen for almost a year, I became aware that the sound company owner made a lot more money and had a lot longer career than the sound mixer. And so, when I had the opportunity, I started my own companies, Modular Sound Reinforcement and Joe's Sound and Salami.

RE/P: What groups and/or performers have you done sound for?
RG: Currently, I am working regularly with Judy Collins, Wynton Marsalis, George Carlin, and we've just started a re-
Continued on page 58

Laurel Cash is a free-lance writer based in Los Angeles.

Chuck Berry onstage with a Modular monitor system at the Rock-a-Rama, Philadelphia, June 1987. The monitors, made by Robyn Gately's company Joe's Sound & Salami, are rectangular instead of a wedge shape, which allows the monitor to be pointed where it's needed, instead of a fixed angle with a wedged-shaped monitor.

Paul Dalen began his live sound career in his band days. He got into it full time when the band broke up, and has mixed for a wide variety of performers. Now in his mid-30s, Dalen is well traveled doing concert sound. He has been to Europe half a dozen times and to Japan twice. This interview was conducted just before he left for another European tour.

RE/P: How did you get started mixing live sound?
PD: My first experiences with live sound came when I was performing with a band. I was the one responsible for keeping up on new developments and determining what would be cost-effective as we expanded our own particular sound system. When the band broke up, I needed a way to earn a living, so I just continued on that same path. Shortly thereafter, I was hired by a group called Morningstar. CBS Records had recently signed this act, and they were opening for the group Styx. That was my first real experience with live concert sound.

RE/P: Were you mixing house?
PD: Yes, I was mixing house sound on that.

RE/P: Do you prefer mixing house to monitor?
PD: I've found it a lot more rewarding to do front house sound. The satisfaction you get helping the performers feel comfortable on stage by mixing monitors is certainly important, it's certainly an integral part of the overall show, but I just enjoy doing house more.

Continued on page 63

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House sound system for Judy Collins' and Arlo Guthrie's performances at Carnegie Hall. Note that the left-hand mid-high cabinet wears lenses to increase the dispersion of the high and super-high horns, while the right-hand box has been tilted at a 30° angle to cover the balconies.

Modular stacks at the Jambalaya Jam in Philadelphia. A 2-box, 4-way system uses the Joe's Slide-Align to align the high and super-high horns to the 12-inch speakers in the same cabinet.

lationship with the band Cinderella, which is a heavy metal act from south Jersey.

RE/P: *Do you normally mix house or monitors?*
RG: My preference is for house. I will do monitors where necessary, but I prefer the freedom of doing the house.

RE/P: *Why?*
RG: I like to be like the conductor in an orchestra. I consider a sound man to be very similar to the conductor in that he is forever shaping things. A conductor does not actually play any instrument, but he directs how the rest of the orchestra is going to interact. His is the last chance to get everything right before the audience hears it.

That's really what a sound mixer does. He doesn't control tempo like a concert master or orchestra leader does, but he does direct the volume of the different parts.

RE/P: *Do you have any studio experience?*
RG: Some limited experience. As a company, we did own a studio for almost 10 years, and we came to a decision that the

business return on the studio, and the current marketplace, were not as great as the returns on a sound company. Also, there is not as much freedom in a studio as there is in a sound company. Freedom to mix. Freedom to go in new directions. Studios are very much locked into a competition game, and sound companies at this point are much freer to pursue their own course.

RE/P: What's different for you in mixing for live sound as opposed to mixing for the studio?
RG: I enjoy the pressure of live sound because there is one shot to get it right. My feeling about the studio is that it's boring! Doing the 37th take on something holds no interest for me.

RE/P: There seem to be two types of live mixing engineers: the active ones who tend to keep their hands on the controls, constantly updating the mix; and the passive ones who set their mix in the first couple of songs and then let the band determine the dynamics. Which are you?
RG: I would definitely say that I am the active type. My feeling about passive mixers is that they really are not doing their job, for the simple reason that there is no way a musician standing on stage can properly hear the dynamics being presented to the audience. It is necessary for the sound man to change things as the show goes along, in order to keep things in proper perspective. Musicians are playing in a unique environment surrounded by the other instrumentalists; they sometimes have to change their audio environment in order to make themselves feel comfortable. These are not necessarily the same changes that make the audience feel comfortable.

RE/P: Does this vary from band to band according to their preference?
RG: Some of the acts I work with prefer me to be not more passive, but more static in the mix. A good example of this is Judy Collins. (See Figure 2.) She is looking for the patented Judy Collins sound. Judy may change the dynamics of her voice. Shelton, the piano player, may change the dynamics of the music from orchestra to orchestra or between a band date and a symphony date, but they are always looking for the end result to be the patented Judy Collins sound. It's necessary for me to update many times during the show in order to keep everything in perspective. So, it's still an active mix; but the actual sound is something that I would call almost static.

RE/P: How important is getting a good stage sound to your final mix? Do you have the chance to work with musicians

in developing their stage sound before the tour, or do you let them do whatever makes them feel the most comfortable on stage and fix it in the mix?
RG: I like making an artist feel as comfortable as possible on stage so they can give their best performance. If an artist needs to play a certain way to achieve the sound he wants on stage, I would rather make him or her feel comfortable and go from there.

RE/P: Do you use a lot of limiting?

RG: It depends upon the act. With more rock-oriented acts, I may have difficulties separating the instruments that come down the vocal channels. The more acoustic the act, obviously, the fewer the problems. The use of various devices on individual mic channels is greatly dependent upon the strength and dynamics of the vocalist and what else I have to contend with in the way of leakage. On some artists, I will not use any kind of limiting or effects at all. With others, the use of limited effects can be necessary.

Figure 1. *Stage plan for Wynton Marsalis.*

Table 1. *Microphone plot for Wynton Marsalis.*

Position	Microphone
Announce	Beyer 400
Trumpet	Senn. 441
Piano hi	Sony ECM-50
Piano lo	Sony ECM-50
Bass mic	AKG D-12
Kick	AKG D-12
Snare	Shure SM 57
Rack	Senn. 421
Floor	Senn. 421
Overhead	AKG 451
Sax	Senn 421

Figure 2. *Pop concert setup for Judy Collins.*

Table 2. *Microphone plot for Judy Collins.*

Position	Microphone
Judy's vocal	AKG 451
Piano vocal	AKG 451
Keyboard	AKG 451
Gtr. vocal on stand	AKG 451
Piano lo	Sony ECM-50
Piano hi	Sony ECM-50
DX-7	DI
Jupiter 8	DI
Acoustic gtr. (wireless)	DI
Bass	DI
DX-7 bass	DI
Kick	AKG D-12
Snare	AT M-63
Hi-hat	AT M-63
Rack	Senn. 421
Rack	Senn. 421
Floor	Shure SM 57
Overhead	AKG 451
Drum machine	DI
Sp. vocal	AKG 451

A good example of this is the comparison between Wynton Marsalis and Judy Collins. (See Figures 1 and 2.) Wynton is much more aware of the volume differences that he can present to a microphone and tends to play away from the mic when he is playing his loud passages. For Ms. Collins, the use of some judicious limiting is certainly helpful because there are limits as to how far she can hold the mic away from her mouth. I find that most vocal artists prefer to remain as unaffected as possible and would rather have a mixer with fast hands than have a limiter or something else in the channel.

RE/P: Does the stage leakage interfere with your ability to get a good mix?
RG: It can when dealing with venues. In smaller venues, there are definitely times when the leakage into the vocal microphone, precludes getting "a studio mix."

A loud rock 'n' roll band can, especially in a large club or small sphere, have quite a bit of leakage to the vocal mic. There are times when the mix seems to be what I would call washed out because of all the leakage. There are only a couple of things that can be done in that kind of situation; try dropping as many instruments out of the mix as you can and just live with the leakage, or turn things up absurdly loud.

RE/P: Tell me about your use of effects.
RG: Usually, I am actively involved in the use of effects. The ability to modify sounds by the use of reverbs, gates, flanges, etc., is something that I appreciate. It is one of the areas a sound man can add to the performance. There are certain acts where I am asked not to be as active as I would like to be. Ms. Collins is a good example. I tend to set up the patented sound and leave it alone. But there are other acts

where the use of echo and reverb can be very helpful in both modifying and modernizing the sound.

RE/P: What microphones do you use for vocals?
RG: I tend to look for microphones that are extremely flat. I feel that a microphone that has a mid-range peak built into it—like an SM 58 or Beyer 400—will require equalizing to remove the peak in order to use it in the monitors. My personal preferences run toward something as simple as a Shure SM 57 (because it's flatter than a 58 and is extremely durable, like a 58 is) all the way to using AKG 451s. If I had my preference, I probably would use a

more esoteric mic, like a 451, all the time, but that would not be a realistic way of dealing with live sound.

RE/P: Do you prefer to use active or passive direct boxes?
RG: These days Modular Sound's business thrust is toward festivals. Because we do mostly festivals, not tours, we are constantly changing bands on stage and dealing with unique setups. As a result, our preference is toward passive direct boxes, because they tend to cause less trouble in the widest variety of situations. An active direct box can sound cleaner in some situations, but it also has the potential for causing more trouble.

We recently had three festival bids come into our office specifically stating they would not accept active direct boxes. The reason for this is not a sonic one. It's a matter of not being able to afford the five min-

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> *"(An omnidirectional mic) gives a great sense of the openness to the piano, without much feedback trouble."*

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utes that might be spent tracing down the problems with an active direct box.

A passive direct box presents fewer potential problems than an active direct box and, therefore, helps ensure the festival moves along on time.

RE/P: Some people take an instrument direct and mic the amplifier. Do you do that and why?
RG: There are occasions where that will be done. I do not, as a general rule, use both. For example, when a bass player likes the mids and highs he gets out of his speaker system on stage, but wants to retain something fairly clear in the low end, it is necessary to take a direct feed off the instrument as well. Generally I try to pick one or the other. I am either trying to get a super clean signal, or I am trying to pick up what the instrumentalist is doing on stage with the microphone.

RE/P: How do you deal with acoustic piano? Do you use a pickup, or if not, which microphones do you use?
RG: I probably have one of the more unique points of view when it comes to piano pickup. My preference, believe it or not, is for a pair of Sony omnidirectional lavalier mics. Most people feel an omnidirectional mic has no place in a piano pickup system. I believe it gives a great sense of the openness to the piano, with-

out giving me much feedback trouble.

However, the one or two lavaliers must be correctly placed. The biggest problem I see with micing acoustic pianos is the lack of understanding microphone placement. It is the kind of thing that is best demonstrated by putting your head inside the open piano and thoroughly exploring the entire soundboard—while someone is playing the piano. You'll find there are certain points that are much louder than others.

RE/P: Have you had to deal with interfacing MIDI and do you see this as something that will affect house mixes in the future?
RG: As a live sound mixer, I haven't been asked to deal with MIDI yet. I have had musicians start to ask me about it, and I can see this being something that could be an important part of my mix in 10 years. I foresee the use of MIDI, interacting with sound systems, as one of the next great fences that we will cross.

RE/P: Is there a particular priority you give to the blend or mix?

RG: I think mixes should be based on some common principles. There are tremendous numbers of live sound engineers who are not aware they should be hearing the mix in a certain perspective. It has become fairly obvious over the last 10 or 15 years that a live mix generally consists of: the lead vocals first with the snare drum directly behind it, but not at the same level as the vocals; the rhythm section, including the kick drum, bass guitar and rhythm guitar, which should all be on a plane behind the snare drum; the rest of the instruments should come directly behind that; and the lead instrument(s) levels, at about the lead vocal level, when present.

RE/P: Let's talk about loudspeakers. How do you feel about horn-loaded vs. bass reflex systems?

RG: You are asking someone who dislikes speaker systems that have a sound of their own unless the music is specifically slated for that sound. Heavy metal bands are often helped by horn-loaded sound systems. The kick drum that kills your chest is something they're specifically looking for and that is really caused by a tremendous peak in the low end that resonates with the kick drum. It works very well for specific types of music, but I prefer something that is much flatter and much less colorful, such as bass reflex-type systems. I also prefer totally aligned systems because I would rather deal with a system that has the least amount of coloration. If I want the sound colored, I want to be able to add it or subtract it myself, not fight against it in the mix.

RE/P: During the mid-70s, bass reflex was considered short throw and horn-loaded was considered long throw. Do you feel that is still true?

RG: It seems that certain manufacturers are addressing the problems and, as a result, there will be some tremendous changes made in the concept of speaker cabinet design over the next 10 years. I believe that bass reflex cabinets have finally come into their own and that people have started to realize what can be achieved through the use of line arrays, etc. The actual use of the bass reflex cabinet to direct the speaker sound is something that is just starting to be explored.

RE/P: In arena systems, there seem to be two schools of thought: the Clair Brothers' way of hanging speakers in the round even if the band is facing only one direction, and the Showco philosophy of left-right stacks with as tight a point source as possi-

ble. Which school of thought do you agree with and why?

RG: The "speakers-in-the-round" point of view suggests if you point a speaker everywhere, everyone will be able to hear the sound. This is the sonic equivalent of pouring pebbles into a pond. Yes, there will be waves everywhere, but you can't pick any of them out. To me, the concept of point-source arrays creates less reverb clutter because the ear is able to locate the speaker much more cleanly than it can in the round system.

RE/P: Do you mix differently for different rooms?

RG: There is no question that different rooms require different mixing perspectives. A highly reverberant room tends to have a mix that has a much greater dynamic range than a room that is very

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"I think mixes should be based on some common principles."

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dead. The lead instruments (vocals and snare for instance) need to be higher in the mix, compared to the rhythm instruments, in a highly reverberant room.

RE/P: Do you ever have a chance to leave the board and listen at other locations in the hall?

RG: If you get trapped behind the console you may be hearing the mix as someone in the audience hears it, but from only one place. The mix can change drastically as one travels around the room. It's necessary to get at least 30 to 50 feet away from the board, early in the show, to check for problems which can ruin the concert. It is better to get a good sound everywhere than to get a great sound at the board and have it sound like hell in the rest of the hall.

RE/P: What is your favorite venue?

RG: Among the larger halls, I really like the Performing Arts Center in Providence, RI, which is a small hockey rink that seats 8,000 to 10,000 people. I also like Stabler Arena on the campus of Lehigh University in Bethlehem, PA. Among smaller halls, I really enjoy Carnegie Hall because of its natural sound. I also am very pleased by the Terrytown Music Hall in Terrytown, NY, which is more than 100 years old and built entirely of wood. It is a very warm-sounding hall without the appearance of any long reverberation. This means that

everything I want has to be put into the mix, which, of course, gives me the greatest latitude.

RE/P: How do you feel about mixing inside as opposed to outside? Which do you prefer?

RG: That's a tough one. I guess I'd rather deal with a good hall than be outside. If I'm outside, and if I have enough PA, I am able to put more in the mix (without it getting cluttered) because of the lack of natural reverberation.

RE/P: Have you used the new switching or Class-D amplifier technology? If so, what advantages or disadvantages have you found?

RG: The biggest advantage is the weight or lack thereof. The biggest disadvantage, I find, is that they either poop out or shut off from overheating much quicker than conventional amplifiers, when used for lows and highs. In designing systems recently, we have used switching amplifiers for all of our mid-range functions but have gone back to conventional amplifier technology for the extremes of the audio spectrum.

RE/P: As a sound reinforcement company owner, what is it that you look for when you're looking to hire mixers for the road?

RG: First and foremost comes attitude: not on how they approach the road, but their attitude when dealing with performers. One of the biggest problems I have is finding people who live, eat and breathe live sound reinforcement—and, at the same time, are more than willing to help the artists achieve what they want on stage. This requires a good deal of communication. You either have it born into you or you don't. It doesn't matter how well you mix, if artists can't communicate what they want, and you can't achieve what artists want, you're not going to survive very long.

Second is the ability of a mixer to understand the placement of the instruments in a mix. Studying record and concert mixes is time that is well spent for someone who wants to be in this business. Over the years, many styles of mixing have been developed by various engineers, both live and recording; but the styles all come down to some basic rules like the ones I explained earlier.

The ability to adapt your mix to the kind of music being played will determine how successful and broad-based your career is. A sound mixer who is equally at home mixing country, folk, pop and rock is going to have a lot more opportunities than someone who only knows how to mix one type of music.

RE/P: *What other groups and/or performers have you done sound for?*

PD: My clients have included David Sanborn, Cock Robin, Stephanie Mills, the Psychedelic Furs, Tommy Shaw, Shooting Star, Robin Trower and Al Jarreau (on a limited basis).

RE/P: *Do you have any studio experience?*

PD: Not much. I've been on site for the recording of several different records with groups I was working live sound for at the time. Other than that, I was involved in mixing 24-track remote tapes for groups I've worked for. These were mostly live performances that were then used for radio broadcasts.

RE/P: *What's different for you in mixing for live sound as opposed to mixing for studio?*

PD: The immediacy of live. The obvious difference is you do it in real time. You don't have the opportunity to do something several times, go for several different takes, or to fine-tune quite as much. Obviously, through a sound check we get the ability to fine-tune certain things, but in some venues, the environment can change so drastically from an empty hall to a full hall that it's little more than a dress rehearsal.

RE/P: *There seem to be two types of live mixing engineers: the active type and the passive type. Which are you?*

PD: When I first started mixing, I was a

"Certain songs need to be pushed and pulled a little bit."

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really active mixer. I felt that by being active, I was helping the performance. As years have gone by, I found myself doing less and less, although, it really depends on the act and the situation. Even in a given act, it can depend on the song. I've found through discussion with the artist that certain songs, by nature of their performance, need to be pushed and pulled a little bit. Some of the drama or dynamics that are taking place need to be accentuated or helped in some situations.

RE/P: *What mics do you use for vocals?*

PD: Generally, I end up using Shure SM 58s for vocals.

RE/P: *Why?*

PD: They're cheap. They're reliable. They're a known factor. I know what they're going to sound like if they're relatively new and in reasonably good condition. With Al Jarreau, I use a Beyer 88. On some other groups, mostly rock, I've tried an Electro-Voice PL 95, which is the old DS 35, because it has good off-axis response.

RE/P: *What type of loudspeaker setup do you prefer?*

PD: Rarely, in my experience, has the engineer's preference been the major deciding factor. In certain situations I prefer a well-designed and -maintained, horn-loaded speaker system over bass reflex. I've found that it's really difficult to design a top-flight horn-loaded system. I believe that an infinitely baffled or a direct radi-

ating system can work just as well. A lot of it depends on the type of music and the venue.

RE/P: In arena systems, there seem to be two schools of thought. First is the Clair Brothers' way of hanging speakers in the round. The other is the Showco philosophy of left-right stacks with as tight a point source as possible. Which do you agree with and why?
PD: I enjoy mixing in stereo, although a good portion of the crowd can't appreciate stereo imaging. What I prefer to do is have a left and right speaker cluster clone stack, with the hope that both the left and the right are as coherent as possible. Meaning that they're aligned, stacked and flown in such a way that the whole array is cohesive and physically time-aligned by the way they're hung.

RE/P: Have you dealt with electronically aligned speakers?
PD: Yes. I've worked with the Electrotec system with electronic time alignment after the crossover. It really made a huge difference.

RE/P: Tell me about your use of effects.
PD: I try to use effects where they'll help, make sense and fit. There are generally two categories. One is sweetening and helping something that already exists. Then there's the use of effects actually as effects, something that very clearly is an addition to what's happening on stage.

RE/P: Do you use EQ as a room equalizer as well as an effect?
PD: Yes. Every system that I've used so far has had graphic equalizers on the left and right outputs of the system. Also on every input strip to try for fine-tuning each sound I get from the stage.

RE/P: How do you deal with acoustic piano? Do you use a pickup?
PD: I've used pickups and I've used microphones.

RE/P: What microphones?
PD: I've had really good luck with a pair of AKG 414s on an acoustic piano. I also sometimes use the C-ducer in conjunction with a microphone.

RE/P: Either using microphones only or the mic/pickup combination, where would you place them?
PD: Let me tell you about the finest piano sound I have achieved in a rock band situation—where there was a lot of stage volume. I had a C-ducer on the bottom of the piano's sound board. The entire bottom of this Yamaha C-3 baby grand was covered with plexiglass. Mounted on a piece of

Sonex inside the lid, I had a PZM. I went to great lengths to isolate the sound acoustically inside the box with the lid down. In addition to the Sonex on the lid of the piano, I had weather stripping mounted all the way around the edge of the piano so that, when the lid closed, the lid was isolated from the box. As a result, you didn't get a lot of rumble, but I did get an incredibly natural piano sound.

RE/P: Which do you prefer, active or passive direct boxes?
PD: I've had good luck using active direct boxes. They sound fine. As a rule of thumb, I ask for a Countryman direct box because I've never had a problem with them.

RE/P: Is there a particular console layout you use when mixing?
PD: Yes. In dealing with a drum kit, a set of vocals or multiple instrumentalists, meaning if there are two or three guitar

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"I won't allow myself to get trapped behind the console."

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players, I do it to correspond visually. Regardless of where the guitar players go on the console, I have them grouped so they are visually oriented as I'm looking from stage right to stage left.

RE/P: What is your favorite console?
PD: I'd have to say that the Gamble is my favorite. It's the most transparent desk I've ever worked with. As for features, I think the Soundcraft Series 4 is tops.

RE/P: Do you mix differently for different rooms?
PD: Yes. You have to mix differently for different rooms. If you don't, you're in trouble. The larger the room and the more dense the reflections, the more you have to accommodate your mix to correspond.

RE/P: Do you ever have a chance to leave the board and listen in other spots in the venue?
PD: I not only have a chance to, I make a point of it.

RE/P: So you don't get trapped behind the console?
PD: I won't allow myself to get trapped behind the console. I tend to do most of my walking during the sound check. I'll get a mix set up and then start walking

around the venue. During the show, I do a little bit. Unless, of course, it's a general admission crowd, in which case I do get trapped by the logistics involved. More often than not, during the show I rely on my systems engineer, the engineer who is provided by the sound company, to be my walking ears.

RE/P: How important is getting a good stage sound to your final mix?
PD: The larger the room, the less important it is. If the monitors are of bad or uneven tonal quality, and are very loud, then it can be difficult to deal with. If you have a bad mix bleeding back to a vocal mic... you have to deal with it.

RE/P: What words of wisdom would you like to pass on to engineers just starting out?
PD: There is a huge amount of politics involved in getting and maintaining work. Some of the mixing jobs that I've had have come not only by what I hope is a high degree of competence, but also because of personal associations.

RE/P: Another case of who you know will get you a job, and what you know will keep it?
PD: Yeah, who you know. A tour manager you know you can get along with. A tour manager who knows I know how to deal with artists who have what could be loosely described as an artistic temperament.

RE/P: How does a newcomer go about getting in?
PD: The normal procedure would be for an engineer to be hired by a sound company and go out as the third or fourth person on a PA crew, and then work up. If you are lucky, you move up to monitor engineer, and then possibly in-house engineer. I was hired right off the bat as a house sound engineer. Through a series of good luck, of being in the right place at the right time and knowing some influential people, I was put in some positions when there were other, more qualified engineers available. The rest, as they say, is history.

RE/P

Facility Profile:
Target Productions

By Paul D. Lehrman
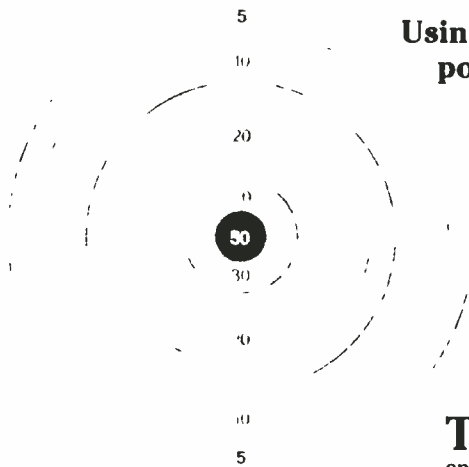
Using the Synclavier allows this video post facility to provide greater flexibility and lower costs for its clients.

The number of options open to the audio engineer designing a post-production studio has increased dramatically. Digital tape, hard disk recording, sampling, MIDI and dedicated workstations are some of the choices that did not exist just a couple of years ago and are now available, often in competing formats.

Target Productions is a new, state-of-the-art video post-production facility in Boston, that was designed from the ground up with high-quality audio in mind. The designers' choice for a do-it-all system for audio editing and mix-to-picture was New England Digital's Synclavier, incorporating both RAM-based sampling with hard-disk storage and recording to hard disk.

Target is located in Charlestown, a residential and industrial section of Boston, just across the river from downtown, in a rehabilitated candy factory known as the Schrafft Center. The facility was started by Chet Collier, a veteran television producer

Paul D. Lehrman is a Boston-based electronic musician, producer, free-lance writer and a regular contributor to RE/P.

and former executive vice president of Metromedia.

Helping him out was Peter Fiedler, a musician and video producer who also happens to be the son of the late Boston Pops maestro Arthur Fiedler.

"It was Chet's perception of what Boston needed," says Fiedler, who now serves as operations manager. "By providing everything under one roof, you'd attract everyone to it."

It was a unique approach in the market. The facility opened in the fall of 1986 with two multiformat video editing suites equipped with seven Ampex VPR-3 1-inch video recorders, dual-channel ADO, Abekas A62 digital hard-disk video storage and CMX editors, a graphics studio with two Quantel Paintboxes and a 25' x40' soundstage. The price tag was about $2.5 million.

"We originally planned on putting all our financial eggs into graphics and editing," Fiedler says, "but we realized that in today's market, audio is no longer just the noise on the other side of the picture. People want better audio quality."

So after a year, in September 1987, Tar-

get celebrated its anniversary by spending another quarter-million or so on the Synclavier, accompanied by a Neotek Elan console, a Studer 24-track analog tape deck and a comprehensive effects rack. The studio is still growing, both physically and in terms of its client base, and whether the Synclavier is an unqualified success is still not certain. But Target is enthusiastic about its chances, and has hired engineer/musician Jeff Largent to run the sound room.

Directly or indirectly, Largent has had several years of Synclavier experience. He used to work at a conventional studio located upstairs from one of the very first Synclavier installations in New England, which didn't do very well; he recalls that they were always coming up for help and extra tracks. But he was intrigued enough to apply for the job when it was announced, and then to take New England Digital's course in running the system before going on staff in August 1987.

Why a Synclavier?
The decision to use a Synclavier was initially based on a number of factors.

"It was partly an image thing," Fiedler says. "We wanted to be unique. Having this system gives you a bit of clout in the industry. But we also had a need to do post-production audio faster than we could before. New England Digital came to us and said they could fill that need."

Largent says: "We looked at the Fairlight, the Sony PCM-3324 and the Lexicon Opus. There was no linear-track disk capability for the Fairlight at the time; and for what it would have cost to provide sound-effects positioning capability for the PCM-3324, we could get a far more flexible system with the Synclavier.

"We all liked the Opus, but it was not yet in production and we needed to be up and running long before it would be available. NED offered polyphonic sampling memory, keyboard control and a linear-recording disk system in one package."

After the decision was made, says Fiedler, the room was designed around the Synclavier. They contracted an architectural firm, Symmes, Maini and McKee, which did the design in conjunction with Bob Peirce, Target's engineering manager, and Steve Blake, a local studio designer.

Although the room is not huge, there's a spacious feel to it, thanks in part to its being built on two levels. Plus, in true video facility fashion, the main hardware and tape machines are in a glassed-off side room.

"It's designed to give the client and operator a comfortable workspace," says Fiedler.

Using the Synclavier
The operating position of the Synclavier consists of a piano-style keyboard, with some 128 buttons and a large LED readout above the keys, a computer terminal with a video display and a detachable QWERTY keyboard. The device actually comprises two different systems that can readily interchange data, but in large part operate independently.

The Direct-to-Disk section is a linear recording system with a variable sampling rate up to 100kHz, using two 80Mbyte double-sided hard disks for storage, with a Fujitsu tape-streamer backup. It records four tracks of audio, like a conventional 4-track tape deck, but with far more versatility. Each track can be accessed and cued independently or in tandem with any other, and tracks can also be bounced to each other, with or without delay added, all entirely in the digital domain. The maximum storage time is about 12½ minutes per track, at a sampling rate of 50kHz.

The other half is the polyphonic sampling system. Here the sounds get loaded into RAM for real-time manipulation, and then are stored on a separate 160MByte Winchester hard disk system equipped with a Kennedy tape drive backup. The sampler allows 32 voices, again using a

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Target Productions' multiformat video editing room.

Target Productions' video machine room.

Photo by Richard Mandelkorn

variable sampling rate of up to 100kHz, and has 16MBytes of RAM.

While the Direct-to-Disk system allows greater storage, the "poly" system provides greater flexibility over each sound.

"If I have a sound effect in the poly memory and I want to change the pitch, I just hit a different key," Largent explains. "I don't have that kind of sampling rate control over individual cues in the Direct-to-Disk section—although I can change the pitch on an entire project. However, if I want to do a 30-second voice-over, and I have 10 takes, there isn't enough poly memory to hold all that. With Direct-to-Disk, I can assemble that voice-over by just specifying individual takes or parts of takes at specific times.

"There is no mixing automation in the Direct-to-Disk system. If we want to fade a track, and it's too big to put into sampler RAM, we have to do it when we actually record it onto the disk. Or else we can fake it by putting a sine wave in RAM with an envelope on it, then run the Direct-to-Disk track into a Kepex and have it follow the envelope of the sine wave."

The sampler editing functions are very versatile. Sounds can be mixed, crossfaded, looped, lengthened, pitch changed, and cut and pasted non-destructively, with the visual display providing a great deal of information.

"To the client, that's the most obvious advantage of the system," Fiedler says. Tying all of the various parts together is a 200-track sequencer, which is part of the system software. A single track on the sequencer can contain an edit list for all four of the Direct-to-Disk tracks. The edit list can be assembled using the "Audio Event Editor," a new software front end for the sequencer, which uses windows and a mouse for assembling sound. A sequencer track can also contain note-on/note-off/key number events for a particular sample in memory, or one channel of MIDI data.

Assembling an effects track using the edit list is simple. "We pull the effects off our library CDs and record them into the sample memory," says Largent, "then we'll type in the numbers on the edit list, or assign them to different notes on the keyboard and fly them in as the picture rolls.

"The limitations inherent in this system are not the same as the limitations in analog tape. If you want more time on an analog system, you just put on another reel. With this, you have to back up the disks, erase them and start fresh. You also have to spend a lot of time adjusting the sounds as they get loaded in—trimming, looping and editing. Right now, with the amount of memory we have in this system, it's impossible to have our entire effects library on line. We're essentially cre-

ating a new database for each project."

The system was originally specified with 8Mbytes of RAM, but it soon became evident that wouldn't be enough. Bob Peirce explains: "It was naivete on our part. We saw other systems running eight megs that seemed to be doing the job we wanted to do. It turns out our stuff was not the same as the others, but there was no way we could have foretold that. Audio involves a lot of gesturing in free space."

"We had to make some difficult decisions based on new and somewhat untested technology," says Largent. "I realized while taking the course at NED that we would need more RAM. It's very easy to underestimate your needs when you're buying one of these. There wasn't really enough probing on the manufacturer's part."

Some more consultation with NED resulted in the purchase of an additional 8Mbytes, which, according to Largent, "turned the whole system completely around. I can make things happen now that just weren't possible before.

"The stuff's not cheap, and if it were up to me I'd buy the biggest system available, but you have to find a balance between what you need and what you can afford. The trick is understanding how these new digital systems will work in your particular situation. We seem to have reached a happy medium.

"Of course," he adds, "we also had an initial budget that had to cover everything else in the room. If we had started with 16 megs, there wouldn't have been any money left over, and we would have ended up with a 1-inch 8-track and no effects."

There was also the question of using what was available with the greatest possible efficiency. "At first," says Largent, "I was green, and sampled everything at 50kHz, but I ran out of room really fast. Now we have more memory and hard disk storage, so there's more room, but I still run out occasionally and have to excuse myself and kick some samples down to 25kHz."

Dealing with the outside world

The Synclavier has a total of 14 audio outputs: an output for each of the four Direct-to-Disk tracks, eight outputs from the sampler and a stereo pair containing a mix of the sample outputs.

The console in the sound room is a 30-input Neotek Elan. Size and versatility were two of the primary reasons for selecting it.

"We had only a small space to work with," says Largent, "and we needed as many as 40 inputs for everything. So it was either a 9-foot-long 50-input frame, or this.

Peter Fiedler, operations manager (left), and Jeff Largent, sound designer, in Target's audio post-production suite.

Target's audio post-production room features a Synclavier with 16Mbytes of RAM, a 30-input Neotek Elan console and various outboard effects.

The Neotek has dual-line architecture, which means I can dump from a monitor pot directly to the mix bus, which gives me effectively 60 inputs. I have the tape returns come up on monitors and use the main faders for the Synclavier output."

The console is situated backward in the room, facing away from the video and audio monitors. By swivelling the chair, the engineer moves back and forth between the console and the Synclavier.

"We originally wanted them side by side," says Bob Peirce, "but however we did it, the engineer wouldn't be in the sweet spot." Because the client sits behind the engineer, in order to see him, the engineer has to turn around anyway.

"It's useful to face the client when you mix," says Peirce, "so you can see the nuances of expression on his face: raised eyebrows, smiles and frowns all make a difference. On the other hand, when you're hitting effects from the keyboard, it's important to see the screen."

The audio equipment in the machine room next door includes a Studer 24-track deck, which is used as a backup if they run out of tracks or RAM, and for longer

programs, like 60-minute documentaries.

"We tell our account executives to refer to it as the 'analog extension of our Direct-to-Disk system,'" Largent says. There's also a Sony analog 2-track with a center time code track, but no digital mastering equipment. Because they mostly print direct to videotape, it is not needed.

There is some signal filtering available within the Synclavier, but Largent admits, "I haven't even played with it—since everything goes through the console anyway, it has all I need. We use the sampled sounds almost 'as is'—we may manipulate them in time, but that's about all."

An effects rack sits underneath the Synclavier console housing several Valley Kepexes, a Klark-Teknik reverb and graphic EQ, Lexicon PCM-70 and PCM-47 reverb/delays, and an Aphex Exciter.

Although the audio room is too busy and expensive for use as a music composition studio, having some musical hardware comes in handy when editing and assembling music tracks. Because the Synclavier was ordered without any synthesizing capabilities (various modules are available, including FM and resynthesis, but the studio didn't need it, according to Largent), that job is handled by a Roland D 50 and a Yamaha TX-81Z module.

"We use the synths when we need a sound that's not in the sample library," Largent says. "We'll also sometimes augment a library track with a synth line—a new bass sound, a lead line or whatever. We might take a loop out of a library piece, put it in the sampler and make a bed of it. Then, to make it interesting, we'll modulate it up and down into other keys without changing the speed, add a few synth notes to cover the loop point and then put on a string bed with another synth." The Synclavier itself talks MIDI, with four independent outs and one in, and all MIDI commands are under the control of the sequencer.

The machine also talks SMPTE, and making it slave to time code recorded on videotape or audiotape is a fairly painless process. "It locks within one frame of seeing valid code," says Largent, "and it's very forgiving of poor code." An Adams Smith Zeta-Three synchronizer is on hand to take care of some of the code-handling chores.

Client reaction

So far, the reaction has been very positive—almost too positive, in fact.

"When clients find out what they can do with the system," says Largent, "they want to keep doing more and more. In a typical mix, if the preproduction sheet says we have 45 effects cues, we end up using more than 200."

Adds Fiedler, "We have to preprogram not just the projects, but also the clients, to make sure they know what they want to do ahead of time, or else it turns into a party in there."

"It's far less intimidating for clients to see the sounds on a nice graphic display," says Largent. "They feel they can understand it, and do more creatively—like putting in a 'ping' instead of a 'zing.' They also like to hear things as close as possible to the finished mix as the mix is being built. That's a piece of cake with this."

Which all leads to a dilemma for those whose job it is to figure out what a project will cost a client. Although the studio does publish a rate card—which shows Synclavier time going for $150 an hour—Fiedler says, "We think it's generally a mistake to charge per hour, so we do most projects on a bid basis."

"But because it's so easy to make changes, it can create a problem" says Largent, "clients tend to get carried away during the production process—they keep changing their minds and spending more and more time."

Fiedler explains, "Lots of factors go in-

to determining the cost, because there are lots of different tasks involved in doing a project: not just editing the sounds, but finding them and loading and storing them, moving them in and out of RAM and disk." Largent adds, "Flat bids are easier to deal with in video, because the engineers have been doing it for years. It's not so easy here."

When all is said and done, however, even with all the expensive technology, Target sees its audio room as an economic alternative for clients. "We did a movie intro for a local TV station," says Largent. "We charged full book rate, and we were still able to bring it in for 60% of what someone else in town had bid."

"We think of ourselves as the best," says Fiedler, "but we don't want to be seen as the most expensive."

And so far the approach has been working well. Bose Corporation did the audio here for a sales film; Prime Computer has been doing training cassettes; Federal Express did a series of corporate videos; and there's been plenty of activity from radio, cable and independent TV stations.

"We have very good word of mouth," says Fiedler. "People who've come in and used the system have come back."

Support, service, upgrades

The staff at Target praises the Synclavier's "positive uptime," and New England Digital's support for its system. Fiedler says: "They're helping us with upgrades, service and promotion. We had a software problem at one point, and they sent somebody screaming down here on the interstate in two hours with a fix."

Peirce agrees. "NED is committed to holding our hand. They understand the support needs of the pro industry. I like always having someone on the other end of the phone when we call."

Target is very happy with the current Synclavier software release known as version "N," which incorporates the Audio Event Editor. "It's a big improvement over 'M,'" says Peirce. "I think they're being driven by the market—especially the Opus and Boss systems—to do these kind of improvements. I like the fact that everything is in software, it makes it much easier to upgrade."

And more upgrades are on the way, in both software and hardware. Peirce is planning at some point to get a 2-Gigabyte Write-Once optical disk system, which will allow storage of up to about six hours of sounds. "We'll be able to store our entire effects library, and whole music libraries, on one disk," says Largent. The disks are removable, and cost about $500 each. Accompanying that system will be an on-line database for effects, which will allow sounds to be categorized under multiple

headings, and retrieved instantly. Also in the works are increased performance capacities of the Direct-to-Disk system, improved tape backup and possibly additional memory.

In addition, Largent is beginning to sketch out a second audio room—part of the game plan all along—which he thinks may use a second Synclavier with 32Mbytes of sample RAM, but with a disk-recording system from a different manufacturer.

"We're still in the building stage," Peirce

says, "and we're not being driven fast enough to increase capacity just yet. The market will determine which way we grow—whether we'll increase the hi-tech aspects, or just improve the simple parts, like doing ordinary voice-over editing."

While Target Productions is certainly an attractive, impressive place, the people behind it know that glitz is not enough. Says Fiedler, "In the long run, it doesn't matter what you have for equipment; it's who operates it."

RE/P

Dealing with Contracts

By Sarah S. Coleman

Whether you deal in verbal or written contracts, there are certain requirements that determine their validity.

You have completed a job. Now, you want to be paid. Depending on your relationship with the client, you may have sealed the deal with a handshake, requested front money and a purchase order, or negotiated a legal contract.

Whether you realize it or not, each day you are engaging in some form of contract—written or verbal. From a simple verbal contract of ordering a sandwich for a client to signing a lease for your facility, you are engaging in contracts. As a facility owner, it is important to know what is required for a contract to be valid and enforceable, and what *your* rights are if the other party doesn't honor the terms.

A contract, generally, is a promise that the law will enforce. However, it is important to realize that not all promises make a contractual obligation. In the case of recording studios, negotiations may take place on the phone with the studio manager sending out a work order at a later

Sarah S. Coleman is associate editor of RE/P.

date. Under some circumstances, a gratuitous oral promise is enforceable if:
• The promisor (the person making the promise) should realize that the promisee (the person to whom the promise is made) would rely on it.
• The promisee reasonably relies on the promise.
• The promisee suffers harm—harm that can be prevented only by enforcing the promise.

When two people exchange promises, they have an agreement. If the agreement is enforceable, a contract is formed.

There is no set contract size, but four elements must be present for a contract to be valid: an offer, an acceptance, acceptance supported by consideration and the legality of the contract's objective (meaning that the contract cannot require an act that the law declares illegal).

The contract must also be made voluntarily and by two parties with legal capacity to contract. Not all people are legally capable of forming a contract. For example, minors, mentally ill or incompetent individuals, individuals under guardianship,

and intoxicated or drugged individuals are viewed by the law and courts as limited in their ability to contract.

At Universal Recording in Chicago, studio manager Foote Kirkpatrick points out that it is now common for studios to request front money after the negotiating is complete and before the studio begins the project.

"Sometimes negotiating takes 20 seconds, or it can take days," Kirkpatrick says. "For our studio, a purchase order confirms their intention to do their project."

Mutual assent and consideration

Two other essential elements for a binding contract are *consideration* and *mutual assent*.

The consideration factor is when two people exchange promises, each promise must have some legal value that must be given in exchange for and that must induce the giving of the other promise.

For example, a state fair coordinator promises to pay ABC Audio $25,000 on the company's 25th anniversary. This would not be an enforceable contract be-

cause the company did not give any *consideration* in the contract. However, if the coordinator, mentioned above, promised to give the company $25,000 on the 25th anniversary and the company would assist at the 1988 state fair, it would be an enforceable contract because the ABC company would give consideration by assisting at the state fair.

In the case of *mutual assent*, the courts presume the parties have mutual assent when one party makes an offer that the other party accepts. When an offer and acceptance occur, a contract is formed. However, the law recognizes four areas that violate this concept of mutual assent: *fraud, duress, mistake* and *undue influence*.

Unenforceable contracts

Even if mutual assent and consideration are present, there are four factors that may prevent the contract from being binding. A contract can be unenforceable due to *fraud*, which is intentional misrepresentation. The method that created the mistake is not important; it is the result that is important. Even if the misrepresentation is unintentional, a court may permit a party to rescind a contract because of the mistaken statement. The law expects that parties will conduct themselves in good faith in their commercial dealings. Failure to do so may constitute a *breach of the contract*, entitling the other party to cancel the contract and obtain damages.

If one party forces another to sign a contract under a threat of personal injury, the contract will be unenforceable because of *duress*. The courts have not developed a clear rule or test for duress, but recently, duress has been expanded to include any type of bad faith or improper pressure in bargaining. Also, if one party lacks knowledge or economic power and the stronger or more knowledgeable party gains an excessively large economic advantage, the contract might be unenforceable on the basis of duress.

To throw out a contract on the basis of *mistake* requires two situations. The first is a mistake resulting from ambiguity in the negotiation of the transaction. The second is a mistake as to a material fact that induced the making of the contractual promise.

While negotiating a contract, the two parties may honestly interpret different meanings. If the court agrees with this premise, it will rule there was no mutual agreement, and as a result there was no contract.

If there is a material fact mistake, the mistake must be one regarding a present or past fact. A mistake regarding a future event is not ground for relief from the ob-

ligations of a contractual promise. If one party is mistaken due to carelessness, it is not likely that the court would relieve a party from the contract obligations. However, if the mistake is an "honest" one, and it would be unfair to enforce the contract against the mistaken party because of the misunderstanding, courts may refuse to enforce the contract on the ground of "mistake."

Undue influence is defined as one party having an advantage over the other party in bargaining ability. For example,

if an agent requests that a client sign a contract to perform at a concert that he is promoting at an unreasonably low fee, this is not enforceable because of the undue influence he has as a result of their relationship.

Concert sound contracts

Although the concert sound business is a little more than two decades old, the industry has matured a great deal in the way its business is operated. Contracts now play an important and necessary role in

the operations ranging from a small, local "Sunday in the Park" to a mega-event such as Live Aid.

Sound system rental companies are involved in global events that are observed through the media by millions of people around the world. Recent examples of these mega-events include the visit of Pope John Paul II; presidential speaking engagements; political conventions; entertainment events such as Live Aid and Liberty Weekend. Before a contract is drawn up for a live sound event such as this, a long period of negotiating takes place.

"People who want to work together are usually willing to negotiate," says Wil Sharpe, president of Showco, the Dallas-based sound reinforcement company.

The negotiations are merely a series of offers and counteroffers. Once the agreement is set, then a contract is written.

Some· of the basic points in a concert sound contract will include:
• The agreement.
• Personnel.
• Compensation (schedule, deposit, amount, cancellation fee, etc.).
• Liquidated damages.
• Insurance.
• Security.
• Prior agreement.
• Arbitration (state where legal operations would take place).
• Attorney fees and costs.
• Non-agency.
• Successor.
• Partial invalidity.
• Venue.
• Modifications.
• Equipment list/crew.
• Crew accommodations.

In theory, contracts are meant to protect reasonable expectations. Although contracts may appear complex and cumbersome, they are an important tool to secure your operation.

If you do not have the resources to hire a contract lawyer or entertainment lawyer to draw up a framework for a contract, one Southern California studio owner suggests collecting contracts from various studios to write your own. Once you have a basic framework for a contract, adaptations can be written for particular jobs.

Your business needs attention in many areas. Legal and financial security is one area in which you cannot afford to make mistakes. Contracts can provide this security before and after the job is done. **RE/P**

Contract terms

Acceptance: The actual or implied receipt and retention of whatever is tendered or offered. The acceptance of an offer is the assent to an offer that is requisite to the formation of a contract. It is either expressed or evidenced by circumstances from which such assent may be implied.

Bilateral contract: Both parties make a promise in a contract.

Duress: Overpowering of the will of a person by force.

Executed: When all the parties to the contract have fulfilled all the legal obligations created by the contract.

Executory: Until all legal obligations have been fulfilled, the contract is executory.

Express: The promise or promises are stated in words.

Implied: The promise or promises are not stated directly but are implied by or deduced from the circumstances, the general language, or the conduct of the parties.

Fraud: An intentional misrepresentation of a material fact made for the purpose of inducing another, in reliance upon the misrepresentation, to part with something of value. (Often confused with misrepresentation.)

Guarantor: A person who promises to answer for the debt, default of miscarriage of another.

Indemnify: To hold harmless against loss or damage.

Misrepresentation: Creating in the mind of another an impression not in accordance with the facts, that person is guilty of misrepresentation. (Often confused with fraud.)

Offer: A proposal by one person to another that is intended of itself to create legal relations on acceptance by the person to whom it is made.

Offeror: A person who makes an offer.

Offeree: A person to whom an offer is made.

Promissory Estoppel: When a promisor makes a promise that a reasonable person would expect to induce the promisee, in justifiable reliance thereon, to take some definite and substantial action, the courts will enforce the promise even if no consideration supports it.

Quasi-contract: Obligations based on promises.

Rescission: The injured party may return what he has received and recover what he has given in performance of the contract or its value.

Revocation: The offeree's knowledge that the offeror can no longer perform the promise in his offer has been held as equivalent to notice of revocation.

Specific performance: Performance of a contract precisely as agreed upon; the remedy that arose in equity law to compel the defendant to do what he agreed to do.

Surety: One who by accessory agreement called a contract of suretyship binds himself with another, called the principal, for the performance of an obligation in respect to which such other person is already bound and primarily liable for such performance.

Unilateral contract: Only one of the contracting parties makes a promise in a contract.

Unenforceable contract: Satisfies the basic requirements for a valid contract, but the courts will not enforce it because of some statutory requirement or some rule of law.

Undue Influence: A form of coercion. (The courts will not grant relief on this ground unless there is a difference in the bargaining ability of the parties.)

Voidable contract: Binds one of the parties to the contract but gives the other party the right to withdraw from the contract.

the time needed to arrive at the steady-state response.

The frequency response the ear perceives at a typical audience location therefore differs significantly from both the far-field, steady-state response and the near-field, direct response of the system. Equalizing for a flat, steady-state response causes the system to sound excessively bright. Equalizing for flat response in the near field causes it to sound bottom-heavy.

To arrive at a flat-sounding system, you must measure and equalize the weighted combination of direct and reflected sound contained within the Haas window. This combined response can be revealed only by a measurement instrument that employs some form of selective time windowing.

Acoustical measurement techniques

The history of audio has seen the development of a number of techniques for measuring loudspeaker systems.

The earliest methodologies involved a swept sinewave input and RMS amplitude detection of the measurement microphone output. Evolving from a simple meter display of amplitude data (with man-

ual plotting) to the use of a motor-driven, strip-chart recorder, the technique provided fine frequency resolution but virtually no immunity to extraneous noise. In theory, the technique can be used to gather phase data, but the methods for doing so (oscilloscope X-Y display or phase meter, for example) are rather primitive and limited. The basic sinewave sweep is still used today, but, because it offers no time selectivity, it is generally employed only in acoustical laboratories with anechoic environments.

The first practical measurement technique for installed sound systems was ⅓-octave analysis with pink noise excitation (such an instrument is called a real-time analyzer or RTA). Being relatively inexpensive and simple to implement, this technique is extremely common in everyday audio practice; a recent elaboration extends it to dual-channel use. For the operator, the RTA's greatest advantages are that it incorporates a kind of rough, built-in curve smoothing (being a low-resolution method) and is faster than sinewave sweeps (although averaging must be used).

It is also true, however, that the inherently limited frequency resolution of the technique carries the disadvantages previously

discussed. Moreover, it offers no time selectivity and cannot gather phase data. The RTA, either single- or dual-channel, gathers and displays the steady-state frequency response of the system under test.

Time delay spectrometry (TDS) and its descendant, time-energy-frequency (TEF) analysis, offered the audio world the first practical methodology for accurately measuring the response of an installed loudspeaker system. Developed by Richard Heyser, TEF employs sinewave sweep excitation with a tracking bandpass filter that precedes the detector. The filter sweep is delayed relative to the excitation sine sweep to compensate for propagation delay in the space, affording both time windowing and a measure of noise immunity.

TEF gathers frequency and phase response data with very high resolution; the resolution is dependent upon capture time, however, and very long sweeps are required to achieve fine frequency resolution. Aside from the operator's judgment, the technique offers no direct way to identify spurious data objectively.

The most recent developments in acoustical testing employ dual-channel FFT (Fast Fourier Transform) analysis. The measure-

ment mode, referred to variously as "transfer function" or "cross spectrum" computation, in effect compares the output signal of the system under test with its input signal. The method uses a known excitation signal (normally pseudorandom noise or impulses, although a swept sine wave also may be used) with the FFT algorithm to derive frequency response estimations within any selected time window.

It offers fine amplitude resolution (although a fair amount of computing power is required) and easily gathers phase data.

Time averaging may be used to enhance the noise immunity of the technique dramatically, but this increases the capture time somewhat. In the absence of extraneous noise, only a 1-second capture time is required to achieve 1Hz resolution across the entire audio band. It is, theoretically, impossible to improve on this measurement speed.

Source-independent measurement (SIM, developed by co-author John Meyer) is an extension of dual-channel FFT analysis, which uses the program material itself (voice or music) as the excitation signal. Because the input signal is taken as the measurement reference, its spectral variations are eliminated computationally to yield the system response within any selected time window. The greatest advantage of the SIM technique in acoustical testing is it permits equalization of a sound system during actual operation (such as in a concert). This allows continuous monitoring and correction of response variations because of absorption by the audience, changes in temperature, and changing humidity. (These can be very significant; see "The Speed of Sound" on page 38 in the April issue.)

With music as the test signal, time averaging must be employed, resulting in a measurement cycle of about two minutes. (Preconcert tests may be made with pseudorandom noise or impulses, however, yielding a 1-second capture time for 1Hz resolution.) Excellent noise immunity is ensured by both the averaging process and the gathering of coherence data.

Dual-channel FFT analysis employing a random input with time averaging may be shown mathematically to offer the best linear fit of any current measurement technique when the system under test exhibits non-linearities (as all loudspeaker systems do, to some degree). Because program material is a close approximation of a random excitation signal, and the technique permits gathering response data during actual operating conditions, SIM yields more accurate and reliable data than any other current method.

We foresee a time, not far in the future, when concert sound systems will be linearized by continuous, real-time, in-concert testing apparatus controlling automated parametric equalization circuitry. The groundwork for the development of such a system is already being laid in field research: in fixed installations, in touring reinforcement, as well as in laboratory R&D. Once implemented, automated correction will bring a high degree of stability and predictability to practical loudspeaker installations.

The benefits of this development for the users of sound reinforcement systems cannot be overstated. Testing and equalization will occur entirely in the background, without intruding upon the creative process. Moreover, the resulting degree of system linearity, maintained with great stability, will bring to the creative work of sound engineering a level of control that surpasses the state of today's art. We look forward to hearing what the most accomplished of today's artists—and tomorrow's—will achieve with such tools.

RE/P

STUDIO UPDATE
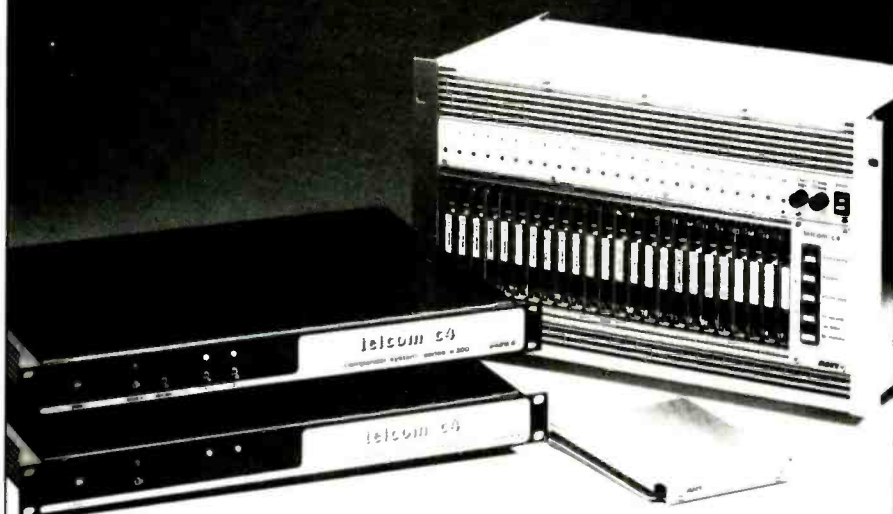
Talkback

Rigging a wireless playback monitor

By Brian McCarty

During the recent film production of Paramount Picture's feature "Police Squad," I was faced with an unusual production problem that was easily solved using two recently developed technologies.

While filming in a Los Angeles-area baseball stadium, it became necessary to play back prerecorded music, for synchronization purposes, a great distance from the camera location. Because of the camera movement and angle, the playback speaker needed to be small and hidden from the camera's view. It was virtually impossible to run a line-level cable to the speaker. In addition, the playback needed

Brian McCarty is an audio technician and a member of IATSE local 695 in Los Angeles.

to be a very loud.

I solved the problem with the use of the newly developed Telex ENG wireless microphone system and the Bose Acoustimass powered loudspeaker system. The output of the Nagra recorder was broadcast, via the Telex unit, several hundred feet to the input of the Bose powered speaker. The Nagra was plugged directly into the Telex transmitter unit, and the input signal to the Bose came from the receiver unit. The speaker then only needed to be attached to ac mains, which were readily available.

The small size of the Bose speaker, combined with the clean output, allowed the scene to be filmed with great precision as

well as minimum setup time, which is so important in feature film work.

Thanks to Brian McCarty for sending in this "Talkback" item.

During a recent job, have you encountered a problem or unusual request that required a unique solution? We would like to share it with the industry. Send it to "Talkback"; if we use it, we'll pay you $50. "Talkback" is a forum for sharing your solutions to difficult production situations other engineers may encounter. In a continuing effort to educate, we believe that this type of information is helpful and will display your professional abilities. This is not a tech tips column; rather, the focus is on solutions to problems—technical or non-technical.

To submit, in 1-2 pages describe the job, what the problem was and what you did to solve the problem. Include any supporting documentation, such as diagrams or photos, that would help explain the situation. If we publish your entry, you and your company will be fully credited.

Send material or inquiries to Michael Fay, Editor, *RE/P*, 8885 Rio San Diego Drive, #107, San Diego, CA 92108.

STUDIO UPDATE

Studio News

Northeast

Trutone Records (Haworth, NJ) has installed several pieces of new equipment in its digital transfer room, including a Sony DAE-1100A digital editor, Harmonia Mundi BW-102 digital processor and Sony DAQ-1000 subcode editor. Ray Janos has joined the staff as digital mastering engineer. *163 Terrace St., Haworth, NJ 07641; 201-385-0940.*

Harmonic Ranch (New York) has initiated the Composer Referral Service as a way for composers to find commissions. Composers submit a demo tape, a resume and completed questionnaire about the work. Producers pay an hourly fee to review tapes. Composers of selected tapes are contacted directly for further negotiations. The service is for all styles of music. *59 Franklin St., New York, NY 10013; 212-966-3141.*

Creative Audio Recording (New York) has named Steve Puccia as rep, concentrating on introducing agency creatives, independent artists and record producers to the facility.

Sound One (New York) has ordered two custom Neve V series consoles as part of its renovation of the former Regent Sound facility.

Editel/NY (New York) has named Cindy Mollo and David Leveen to its editing staff. *244 E. 44th St., New York, NY 10028; 212-867-4600.*

Evergreen Recording (New York) has renovated its Studio A control room, making it twice its previous size, to more comfortably accommodate MIDI and video sessions. Design work was by owner Joel Greenbaum and Al Fierstein of Acoustilog; construction was by Chris Bowman of CHBO Inc. *215 W. 91st St., New York, NY 10024; 212-362-7840.*

Todd-AO East/Glen Glenn Studios (New York) has ordered two more Solid State Logic SL 5000 M series consoles. The consoles will be installed at Todd-AO East (formerly Trans-Audio) in a new facility now being developed on West 54th Street.

Pro Audio/Big Mo Recording (Wheaton, MD) is now offering 24-track studio recording, in addition to its 24-track mobile truck. The 24' x30' x12' studio features hardwood floors, variable acoustics, iso rooms and a 9-foot Kawai grand piano. *11264 Triangle Lane, Wheaton, MD 20902; 301-946-7364.*

Southern California

Record Plant (Los Angeles) has completed a major expansion and has installed a Neve V series 60-input console with GML automation in Studio One. That studio's

SSL 56-input console with Total Recall has been upgraded to the 4000 series G and moved to the Stage L scoring operation on the Paramount movie lot. New to the staff is John van Tongren, the resident synthesist for Stage L. *1032 N. Sycamore, Los Angeles, CA 90038; 213-653-0240.*

Sunset Sound L.A. (Woodland Hills) has installed an SSL G series computer and a Studer A820 2-track recorder. *4836 Queen Victoria Road, Woodland Hills, CA 91364; 818-999-6160.*

Preferred Sound (Woodland Hills) has added a Trident series 80 40×24×24 console in Pacific Sound, its second room. *22700 Margarita Drive, Woodland Hills, CA 91364; 818-883-9733.*

Skip Taylor Recording (Los Angeles) has added a Neve 10-channel Prism EQ rack to complement its SSL 4072 Total Recall console. The console itself has been upgraded with eight modules, bringing its input capacity to 64. In addition, the facility's Lexicon PCM-70s have had their software updated. *506 N. Larchmont Blvd., Los Angeles, CA 90004; 213-467-3515.*

Sunset Sound Factory (Hollywood) has completed a renovation of Studio A. New purchases include: two Studer A80 MkIV 24-track recorders; an Otari DTR-900 digital 32-track; Lexicon PCM 70s; Drawmer gates; dbx 160Xs, 902s, 903s and 904s; Roland SDE-3000s; a Publison Infernal 90; and an AMS 1580S. A customized API mixing desk has been installed with an electronic programmable mule matrix system. *6357 Selma Ave., Hollywood, CA 90028; 213-467-2500.*

Northern California

Melchor Productions (Mountain View) has made two staff additions. Jaquelin Hall-Kallas is production manager, and David R. Seedall is chief engineer. *2415 Charleston Road, Mountain View, CA 94043; 415-961-9300.*

Pacific Mobile Recorders (Carmichael) has remodeled its 24-track remote facility. New equipment includes a Harrison MR-4 36-input console, which complements its Otari MTR-90 24-track recorder. *2616 Garfield Ave., Carmichael, CA 95608; 916-483-2340.*

Independent Sound (San Francisco) has appointed Kathy Braun as its San Francisco representative for commercial work. *2030 Scott St., San Francisco, CA 94115; 415-563-2390.*

SRO (San Francisco) has installed a CMX CASS audio editing system with TimeLine LYNX synchronizers. *1338 Mission St., San Francisco, CA 94103; 415-863-0400.*

Midwest

Pearl Sound Studios Ltd. (Canton, MI) has purchased a Neve 48-input V series console with 32 channels and Audio Kinetics Mastermix automation. *47360 Ford Road, Canton, MI 48187; 313-455-7606.*

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STUDIO UPDATE

TRC Studios (Indianapolis) has installed Diskmix Moving Fader automation to its Trident 80B console in Studio A. In Studio B, two adjoining MIDI suites are being constructed, and the studio is being upgraded to accommodate film and video mixing. Construction was scheduled to be completed in June. *5761 Park Plaza Court, Indianapolis, IN 46220; 317-845-1980.*

Canada

River Audio (Fort Erie, Ontario) has named Ed Stone as chief engineer. *Box 1003, 133 Niagara Blvd., Fort Erie, Ontario, Canada L2A 5N8; 416-871-6230.*

Southeast

Alpha Audio (Richmond, VA) has installed a New England Digital Synclavier and Direct-to-Disk system. *2049 W. Broad St., Richmond, VA 23220-2075; 804-358-3852.*

Strawberry Skys Recording Studio (West Columbia, SC) has installed a Forat MIDI update for its Linn Drum. Other recent purchases include an Oberheim Prommer/Sampler and a Magnavox CD player. *1706 Platt Springs Road, West Columbia, SC 29169; 803-794-9300.*

Southwest

Goodnight Dallas (Dallas) has added a Sony 3324 digital recorder and a Sony PCM 2500 DAT recorder. Other recent additions include a Roland D-50 LA, E-mu Systems Emax, Yamaha DX-7II and an RX-5 drum machine.

Manufacturer announcements

Amek/TAC has installed a 36-input Amek Angela console with 32-track monitoring and an external patchbay at Key West, FL-based Shrimp Boat Sound, owned by Jimmy Buffett.

Soundcraft has announced the following console installations: Oingo Boingo's Danny Elfman, a 36-input TS12 for his home studio; Firesign Theatre's Fred Jones, a 600 patchbay console; Projections Incorporated, Ann Arbor, MI, 36-input TS12; and Backstreet Edit, New York, a 200B.

Sony has installed PCM-3324s in the following facilities: Rock Video International and Interfallactic Recording, New York; Audio Force, New York; Private Music, Los Angeles; Record Plant, New York; and Devonshire Studios, North Hollywood.

Otari has announced the following sales: Sunset Sound, Los Angeles, DTR-900; Pantera Productions, Miami, MTR-90; Shrimp Boat Sound, Key West, FL, MX-80; Mirror Image Studios, Gainesville, FL, MX-80; Manhattan Center Studios, New York, DTR-900 and MX-80; the Apollo Theater, New York, MTR-90, MTR-12 II, three CTM-10s, MX5050B-II and MTR-20-T; and Sound One, New York, six MTR-90s.

Prairie Sun Recording

By Dan Torchia

A technically sophisticated studio in a beautiful rural setting allows clients to concentrate without worrying about urban distractions.

Sonoma County, CA, doesn't seem like the most obvious choice when deciding where to locate a recording studio. The county is an hour's drive north of San Francisco, and any studio locating there has to compete with Bay-area and Marin County facilities for business.

Sonoma County, however, is directly north of Marin County and is in the middle of a development boom, the beneficiary of Marin's no-growth policies. And as the county grows, so does a Sonoma County-based studio: Prairie Sun Recording.

Having recently completed its 11th year of business, and the eighth in its present location, Prairie Sun has carved out a niche for itself as a music studio and has positioned itself for future growth.

Its rural setting has advantages. Located on a converted chicken ranch on the outskirts of Cotati, population 3,500, Prairie Sun promotes a relaxed atmosphere where clients can concentrate on their projects and work in a studio that is as technically sophisticated as those found in the Bay area.

"We care about our clients," says Mark "Mooka" Rennick, Prairie Sun's owner. "We feel lucky to be able to do our work here and not have to commute to the city."

Spread out over a 12-acre complex, Prairie Sun consists of two 24-track rooms, a MIDI room, rehearsal room and lodging

Dan Torchia is staff editor of RE/P.

facilities. Because of the location, the facility depends less on a constantly changing client base and instead has developed long-term relationships with artists from the Bay area and with independent record labels.

Background

Rennick's background in the studio business dates to 1976. He moved to Northern California from Illinois to study and play music. His decision to start a studio stemmed from a recording experience with a band he was in.

"This guy came in with his 2-track system. He was just horrible, really mean and rude, and I said to myself, 'Well, screw this, there's got to be a better way.'"

Prairie Sun began in 1977 as a garage studio in Rennick's house in Cotati. He cut a hole in one of the bedroom walls and installed a 1-inch 8-track machine. In 1979, he traded the 8-track machine for a 2-inch, 15-track machine—a 16-track machine with one of the tracks destroyed.

"Whenever I had somebody with a 16-track tape, I went to a friend's studio and transferred one track," Rennick says.

Prairie Sun's Studio A features a Trident TSM 48×32×80 console, Studer and Ampex tape machines, and Q-Lock synchronization. During a typical session, tracking and overdubs occur in Studio B, with mixing in Studio A.

STUDIO UPDATE

At A Glance

Owners: Mark "Mooka" Rennick and Clifton Buck.
Studio manager: Mark "Mooka" Rennick.
Designer: Gary "Doc" Shaefer.
Chief technician: Gary "Doc" Shaefer.
Assistant engineer: Mark Reyburn.

Studio A
Studio dimensions: 25' ×30'.
Control room dimensions: 10' ×15'.
Console: Trident TSM 48×32×80 automated.
Tape machines: Studer A80 MkIV 24-track, Ampex ATR 102 2-track.
Synchronization: Q-Lock.

Studio B
Studio dimensions: 8' ×10'.
Control room dimensions: 22' ×23'.
Console: Trident Series 80.
Tape machines: Studer A80 MkIII 24-track, Ampex ATR 102 2-track.

Studio C
Eight-track preproduction room featuring a Tascam recorder, Tascam 16×2 board, Emulator and an Apple Macintosh Plus computer with Mark of the Unicorn and Digidesigner software.

Compressor/limiters
1176s (4).
dbx 160Xs (4).
dbx 165.
Publison Stereo Tube Limiter.
Neve stereo limiter.
Altec Tube Limiter.

Signal processors
Publison Infernal 90, Fullmost.
AMS RMX16, DSM-15-80.
Yamaha SPX-90s (4).
EMT Stereo.
Yamaha REV-7.

Eventide 910.
Lexicon PCM 41 (2).
DigiTech 1900 (2).
Lexicon 93.
EXR Aural Exciter.
Aphex Type C.
1,000-square-foot live chamber.

Noise reduction
Dyna-Mites (4).
Drawmers (2).
Kepex (20).

Equalizers
Sontec parametric EQ.
Rane PE-15 (20).
Pultec EQP.
Land PEQ2.

Monitors
UREI 811, 813.
Yamaha NS10Ms.
MDM-4.
Wharfdale.
Diamond.

JBL-4311.
B&W.
Big Red.

Instruments
Kimball grand piano.
Emulator II.
Macintosh linkup.
Prophet 10.
Marshall amps.
Yamaha drums.
Various guitars/basses.

Rates
Studio A: $85/hour; $900 daily lockout; $5,774 weekly lockout.
Studio B: $85/hour; $700 daily lockout; $4,375 weekly lockout.
Studio C: $20/hour.

Address: P.O. Box 7084, Cotati, CA 94928; 707-795-7011.

Participants at a recent mixing session in Studio A: at the board, engineer Steve Fontano (left) and Greg Howe, Shrapnel Records' recording artist. Standing behind the console is Mark Reyburn, Prairie Sun's assistant engineer, and Mark "Mooka" Rennick, Prairie Sun's owner. Prairie Sun has established long-term client relations with several independent record labels.

The move to 24-track occurred in 1978, and the move to the ranch occurred in 1979. Although some of the buildings on the complex were built at the turn of the century and were not usable, others were well constructed. With isolation not a concern—the buildings are far enough apart from each other and there are no close neighbors—Rennick was able to concentrate on equipping the studios with quality gear.

Vigilant maintenance
The constant in Prairie Sun's evolution is that Rennick has bought quality used gear at salvage prices and then refurbished it. Once the gear is refurbished, regular maintenance keeps it in top condition.

"I was lucky enough to work with some maintenance guys very early in my career, and they impressed upon me about the industrial nature of the pro gear," he says. "The stuff is built to be fixed."

Gary "Doc" Shaefer, Prairie Sun's chief technician, has been affiliated with the studio for three years. Also the chief technician for Narada Michael Walden's Tarpan Studios, Shaefer designed Prairie Sun's "LEDE-style" control room. He has recently completed another LEDE-style room for the Beach Boys in Monterey.

Studio A, the facility's mixing room, features a Trident TSM 48/32/80 console with disk-based automation, a Studer A80

STUDIO UPDATE

MkIV 24-track machine and an Ampex ATR 102 2-track machine. Studio B, also a 24-track room, features a Trident Series 80 console, a Studer A80 MkIII 24-track machine and another ATR 102. Both rooms are designed to work in tandem, Rennick says.

"When you come down to the studio, you can track and overdub in B at a very reasonable hourly rate," Rennick says. "When you're ready to mix, you go to the automated room [Studio A].

"We're really trying to cater to out-of-town people. You can come in, stay, rehearse, record the album and get out the door within 21 to 30 days, with a surprisingly low budget and very high quality."

Service business

Although many clients from the Bay area readily make the 1-hour drive to Prairie Sun, Rennick says, others stay in the city for short projects, such as an overdub session. To compensate, Rennick and his staff work twice as hard to serve their clients, believing that when choosing between two technically equal studios, people choose the studio that treats them better.

Rennick credits the recording engineers, including Steve Fontano, Alan Sudduth, Dino Alden, Rickey Sanchez, Steve Counter, Jim Gaines and Tom Flye, as active participants in making Prairie Sun a success. Rennick has developed loyal relationships with all, who, in turn, foster loyal relationships with clients.

Future plans

Future plans include audio/video 48-track lockup for Studio A, scheduled to be up and running this year. Also in the works is another room, similar to Studio A, that will be built off of Studio B, so that there will be two compatible control rooms. The present Control Room B will be turned into an iso booth.

Prairie Sun is also into artist development and music publishing. For Rennick, this is a natural and necessary extension of the studio business, one that studios need to take as the 1990s approach.

"I'm not here just to run a recording studio," he says. "To me, it's not just owning a facility. The question is, what are you doing with the facility? Where are you going with it?"

RE/P

NEW PRODUCTS

Soundtracs FMP mixer

Using the same mono and stereo inputs as other Soundtracs FM models, the FMP includes a subgroup/master module that houses six LED bar graphs for metering the subgroups, masters and solo outputs. It is available in 16-4-2 or 24-4-2 formats. Additional features include individual phantom power selection, separate comprehensive effects return, an assignable oscillator and a built-in talkback microphone.

Circle (101) on Rapid Facts Card

New England Digital graphics work station

For use with the Synclavier and Direct-to-Disk recorder, the work station is a customized Macintosh II that can incorporate two high-resolution, 19-inch monitors. The work station provides crisp color graphics and the ability to run applications software simultaneously. Systems owners who return their present system terminals by Sept. 1 will receive credit toward the purchase of the new system, according to the company.

Circle (102) on Rapid Facts Card

CMS PC MIDI system

The rack-mount unit takes up seven inches of rack space and includes a 10MHz XT motherboard, a 150W power supply, 3.5- and 5.25-inch floppy drives, MIDI interface board, Cakewalk MIDI sequencing software, a 9-inch internal monitor, four expansion slots and a 50/60 keyboard. The company's MIDI interface is compatible with the Roland MPU-401, and is built around the Roland chip set. The unit is also available with a 12-inch external model instead of the internal monitor.

Circle (106) on Rapid Facts Card

Electro-Voice RE45N/D microphone

A neodymium mic, the RE45N/D is a dynamic shotgun mic designed for ENG/EFP applications. The slip-on Warm Grip handle and switchable low-frequency roll-off help reduce wind and handling noise. The Cardiline design delivers unidirectional characteristics with smooth, off-axis response and cardioid pickup pattern at low frequencies.

Circle (103) on Rapid Facts Card

Studer C270 series recorders

The series comes in three versions: ½-inch 2-track, ¼-inch 4-track and ½-inch 8-track. All feature three tape speeds (3¾ips, 7¼ips and 15ips), variable speed built in (−33% to +50%), and an internal monitor speaker and headphone output with a selector for individual, stereo or mono output. Dolby HX Pro and proprietary phase-compensated electronics are standard. Additionally, the 4- and 8-track versions are configured with LED and VU meters in their front panels.

Circle (104) on Rapid Facts Card

NEW PRODUCT

Otari MTR-100A
24-track recorder

The MTR-100A is a digitally controlled, pinchrollerless analog machine that features automated alignment of all record and reproduce parameters, including level, bias, HF, MF and LF record EQ, phase compensation, HF and LF repro EQ, and repro level. Additional features include a tilt-down front panel with an alphanumeric keypad and a backlit LCD display for entering transport and audio alignment parameters. The VU meter display can be remotely configured and swings up for easy viewing during manual alignment.

Circle (100) on Rapid Facts Card

Gold Line/Loft headphone distribution systems

The HPA-1 and HPA-2 are 6-channel active systems that take up one rack space. Each has its own separate input that independently bypasses the master stereo inputs, which allows any combination of master stereo program or separate programs for custom monitor mixing. Model HPA-2 features a high-impedance mic input to allow the control room to communicate with all six channels.

Circle (105) on Rapid Facts Card

Bruel & Kjaer
frequency analyzer

Model 2133 uses digital filtering for real-time frequency analysis of acoustic and vibration signals up to 22kHz, in bands as narrow as 1/24-octave. The unit is menu-driven, and contains enough internal buffer memory to hold more than 1,000 ⅓-octave spectra. Additional storage is provided on a built-in MS DOS-compatible floppy disk.

Circle (107) on Rapid Facts Card

Christie Electric
CASP/1000

CASP/1000 is a universal battery support system that rejuvinates, analyzes and charges batteries, including NiCad, zinc-silver, lead acid or lithium. It accepts inputs from 90Vac to 265Vac, at frequencies from 47Hz to 440Hz, can charge batteries requiring up to 78Vdc and delivers a charge current up to 10A maximum. Three color-coded keys, labeled charge, analyze and stop, control all the functions.

Circle (108) on Rapid Facts Card

Anvil Cases MICS
modular case system

Standing for Modular Interlocking Case System, MICS features cases that convert to workstations, eliminating storage problems after cases are emptied. No tools are required, and work surfaces can withstand a 150-pound load. The system is available in both its ATA-grade cases and MACC line.

Circle (109) on Rapid Facts Card

Sola surge suppressors

Three different styles of power suppressors are available: a power control center, plug strip models with six outlets and duplex plug-in models with two outlets. They are designed to clamp power line transients in less than 5ns for common-mode and less than 1ns for normal-mode.

Circle (111) on Rapid Facts Card

Memory upgrade for
Fluke 9100A, 9000

The HyperTEST algorithm is now available for the interface pod for the 9100A digital test system and the 9000 series micro-system troubleshooter. RAM tests are performed up to 60 times faster while maintaining fault coverage comparable to the present built-in 9000/9100 memory tests. ROM tests are performed approximately five times faster. The enhancement is standard on new equipment, and is available as an upgrade for existing systems.

Circle (110) on Rapid Facts Card

Yamaha MV422
multi-source mixer

The unit allows the selection and mixing of a variety of sources. The MV422 features four mic inputs, each with −20dB, −35dB or −50dB input gain, 2-band EQ, L/L+R/R pan switch, independent monitor level and an effects-send control. The line input section is organized into two source groups, each having its own level and monitor level control.

Circle (112) on Rapid Facts Card

Frazier CAT 38 monitor

CAT stands for Coincident Aligned Transducer design, which means that the monitor has no crossover notch at any angle, and may be used either upright or on its side. The monitor is a 2-way system consisting of an 8-inch woofer in a B4 aligned, tuned enclosure and a 1-inch dome tweeter mounted coaxially with the woofer. Sensitivity is 88dB (1W/1m), and frequency response is 70Hz to 18kHz, ±3dB.

Circle (113) on Rapid Facts Card

AMS Logic 1
digital mixing console

The Logic 1 offers total automation of all functions while having the appearance of a conventional console for ease of use. All functions are under computer monitoring and real-time updating, not just snapshot control. The linear motorized fader system reduces moving parts to one and offers a more natural feel, according to the company. The console has been initially configured with the AudioFile.

Circle (117) on Rapid Facts Card

NEW PRODUCTS

S&R Stylyx console

The Stylyx starts with a basic frame and can be customized to fit individual needs using various modules, including mic/line, stereo line, subgroup, master, meter bridge, rack-mount power supply and automation. The automation package is a soft muting system that can be manually, sequencer- or time code-controlled.

Circle (114) on Rapid Facts Card

JBL 2427 compression driver

The 2427 replaces the 2425 compression driver and 2327 horn adapter by incorporating both into a single unit. Phase cancellations have been minimized by a concentric phasing plug; a machined ring of copper surrounds the pole piece to counteract the inductance of the voice coil at high frequencies. It is available in 8Ω and 16Ω versions.

Circle (115) on Rapid Facts Card

Offbeat Systems scoring tools

The company has introduced three products for computerized music scoring applications. The Streamline Scoring System designs cues, creates superimposed color streamers and plays studio-quality click tracks. Custom Timing Notes keeps comprehensive breakdown notes. Click-Station plans and prepares music cues when video cuing is not required.

Circle (116) on Rapid Facts Card

"Sound FX" library on CD

Available from Associated Production Music, "Sound FX—the Library" is a set of 25 compact discs containing thousands of sound effects that have been recorded, not sampled. Depth, presence, dimension and stereo imaging can be altered, and numerous edit points allow latitude and creative freedom in creating effects. The library also comes with a users' handbook.

Circle (118) on Rapid Facts Card

API 3124/3124M mic pre-amp

Available in a 2- or 4-channel version, the unit contains XLR mic inputs internally selectable for 150Ω or 600Ω and a front panel ¼-inch Hi-Z input at mic level. The output section consists of an XLR balanced output that clips at +28dB, and a ¼-inch balanced output that clips at +22dB. The 3124M contains the features of the 3124 plus a stereo bus, aux bus, mix level control, panning, aux send and an optional insert point for each channel.

Circle (120) on Rapid Facts Card

SWR Engineering SM-400 bass amp

The SM-400 is a rack-mount unit that includes a tube pre-amp, a limiter using field effect transistors, EQ controls using ICs and two individual power amps using discrete, solid-state devices. The amps can be used individually to provide biamp capabilities when used with a built-in adjustable electronic crossover, or they can function as two stereo 200W units.

Circle (121) on Rapid Facts Card

Samplevision digital audio editing software

Developed by Turtle Beach and distributed by Digidesign, the software is a mouse-driven graphic user-interface for the IBM PCs or compatibles. Waveforms can be viewed in any resolution and edited with up to 1/50,000 of a second accuracy. Sounds can be analyzed using 3D FFT frequency analysis and then modified using the program's digital EQ. System requirements are an IBM PC or true compatible with 640K memory, dual floppy drive, graphics adapter and a Roland MPU-401 or compatible MIDI interface. The program is available for the Akai S900 and will soon be available for the E-mu Emax, Roland S-50 and Ensoniq EPS.

Circle (122) on Rapid Facts Card

Soundcraft series 200B/VE console

The console is an 8-input version of the series 200B and comes with a parallel interface for direct connection to a video editor. It can operate in a stand-alone mode or be under control of the video editor. Standard, sweep EQ or stereo input modules may be specified. An LED indicates when the console is under the editor's control.

Circle (125) on Rapid Facts Card

Plastic Reel Corporation VHS slip sleeves

The cardboard boxes come in black and are fully assembled, saving duplicators time and money. They come in 225 per master carton.

Circle (119) on Rapid Facts Card

Ariel SYSid acoustic measurement system

The system consists of the Ariel DSP-16 board, an IBM PC or compatible and SYSid software, developed by AT&T Bell Labs. The system stimulates the system under test using a chirp, impulse or tone. It records, averages and analyzes the response and displays the system's attributes, including transfer function, distortion, impulse reponse and noise floor. Two independent channels can be measured simultaneously, making impedance and ratiometric measurements possible.

Circle (123) on Rapid Facts Card

Panasonic SL-4300, -4700 compact disc players

The SL-4300 is a single-tray player with an access time of less than one second. A 10-key keypad allows up to 20 tracks to be selected in advance. A dedicated rear-panel socket allows multiple players to be connected together for automated series play. The SL-4700 has a removable magazine that holds up to six CDs. Up to 36 "steps" can be programmed, either to a single disc or across all six. Both are designed to be rack-mounted.

Circle (126) on Rapid Facts Card

Alpha Wire Corporation wire, cable and accessories

The company has introduced a cable and accessories for a variety of applications, including video, audio, mic, data and coaxial. The mic cable features a high-density Starquad design that reduces hum and noise to less than 10% of that with conventional two-conductory mic cable. It is designed to perform in an operating temperature range of −20°C to +60°C. It is available with PVC jackets or with rubber.

Circle (128) on Rapid Facts Card

Output option for Kurzweil 250

The Separate Output Option Kit is available for both the 250 and the 250 RMX, and adds 12 direct mono outputs to the two mix outputs on the basic 250. New software with the kit provides improved channel stealing capabilities and the provision for sending MIDI program change commands to external devices on all 16 MIDI channels. It is available as an option on new units, and is available as a retrofit. Units from 1984 or earlier may require a CPU upgrade.

Circle (127) on Rapid Facts Card

Valley International Micro FX series

The series initially contains seven products: compressor, noise gate, de-esser, noise reduction, effects booster and effects attenuator. Each is powered by a remote 9Vac power supply. Using an adapter allows up to three units to be mounted in a 19-inch rack, taking up 1¾ inches.

Circle (124) on Rapid Facts Card

Out Board Electronics motorized fader

The MF100-S is a motorized fader without pulleys, drive belts, slip ring contacts, springs, strings or clutches. It is designed to retrofit most audio and lighting faders, and requires a bidirectional dc-current drive and a 3-wire servo track connection for position feedback to the control system. According to the company, the motor is undetectable in manual operation because the fader's inertia is so low.

Circle (129) on Rapid Facts Card

HME DN100 antenna distribution system

The DN100 allows four HME RX520 switching-diversity receivers to be rack-mounted with only two antennas. The system contains one DN100, an ac adapter and locking clip and eight RG58 BNC to BNC coaxial cables. A specially designed circuit guarantees that there is no signal loss from antenna splitting, according to the company.

Circle (130) on Rapid Facts Card

LETTERS

because their performance is not acceptable when compared to "air condenser shotgun mics"—currently.

Fred Ginsburg replies:

With regard to Bruce Bartlett's recent attempt ["Letters," April issue] to "clarify" several facts stated in my background article on production sound: Mr. Bartlett disagreed with my observations that the current breed of electret condenser microphones (Sennheiser K3U/ME-80, Audio Technica AT-835, Sony ECM-672, etc.) lack the sensitivity and reach offered by RF condenser shotgun mics (such as the Sennheiser MKH416/816, etc.). He strongly believes that electret capsules are capable of the same sensitivity as "air" condensers. Although that may be true from a laboratory standpoint, the point that I was making is that, from a practical standpoint, the condenser offers superior performance and working range over the electrets currently widespread on the market.

As to the question over my use of the terms "transparent" and "proximity" in regard to lavaliers, I must emphasize that these terms are not synonymous with "omnidirectional" vs. "unidirectional."

All of the lavaliers in question are, in fact, omnidirectional electrets. However, there is a substantial difference in how they react to background ambience. Professional mixers universally agree that there is definitely a distinct difference in the sound of a "proximity" lavalier (such as an ECM-50) compared to that of a "transparent" lavalier (such as a Tram, MKE-2 or ECM-77). These differences have nothing to do with pickup pattern, but instead are related to EQ and relative sensitivity.

Unidirectional lavaliers (cardioids such as the Sony ECM-66 and Sennheiser MKE-40) are in a category by themselves and are of fairly recent invention. Because of a number of factors and characteristics, unidirectional lavaliers have not found wide acceptance for theatrical film/video production. Instead, their application tends to be more popular in sound reinforcement and conference recording.

Thank you very much for allowing me to "set the record straight."

RE/P

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